# Quality of Packet Services in UMTS and Heterogeneous Networks - Objective and Subjective Evaluation

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*Publication date:*
2006

*Document Version*
Publisher's PDF, also known as Version of record

# Quality of Packet Services in UMTS and Heterogeneous Networks

## Objective and Subjective Evaluation

by

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A dissertation submitted to the Faculty of Engineering, Science and Medicine
of Aalborg University
in partial fulfilment of the requirements for the degree of
Doctor of Philosophy,
Aalborg, Denmark,
March 2007

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R06-1003
ISSN 0908-1224
ISBN 87-90834-96-8

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# Abstract

In this thesis, the performance investigation of packet services in both a standalone UMTS network and a UMTS network coexisting with other networks is undertaken. The investigations are concerned with the performance of a single link. Apart from objective performance metrics such as goodput and delay, subjective measures of user perceived QoS are also used. A real time emulation platform that uses a combination of simulated and real network protocols has been developed to enable both subjective and objective investigations from the user's perspective.

With regard to a standalone UMTS network, detailed emulation results are given concerning the performance enhancements due to the several mechanisms that control UMTS link layer retransmissions. The results show that the maximum number of retransmissions is the most determinant factor, and setting it to three or more eliminates TCP retransmissions and gives optimal performance (up to 85% improvement in the goodput). The impact of TCP spurious retransmissions and the improvements due to several proposed TCP extensions such as Eifel are also investigated. Eifel leads to 33% improvement in the goodput while DSACK improves the performance by only 14% when out of order delivery is employed at the RLC layer. However, for the delay spike cases, the improvements are less than 9%.

The impact of inter-system handover and associated buffering techniques, including a new technique called intermediate buffering, on different TCP extensions when UMTS coexists with other networks are investigated. Full buffering during handover can lead to the worst performance, as compared with no buffering at all, and intermediate buffering gives the best result. Eifel improves the performance when the handover delay is small and full buffering is employed, but it causes up to 34% degradation in the goodput when

full buffering is not used. SACK has no effect on full buffering and on cases where timeouts occur. However, it leads to a significant drop (up to 30%) in the goodput for the upward handover cases, and up to 11% increase in the goodput for the downward handover cases.

Finally, usability experiments are carried out to see if there is any noticeable trend in user perceived quality of service and if so, if the subjective measures are in line with objective measures of quality. The user's perceived quality ratings are mostly in line with objective quality measures, justifying the use of objective measures for quality of service investigations.

This thesis tries to give a holistic view of packet service performance by considering the impact of different protocol layers and taking the end user into account. Operators and service providers can make use of the results within this thesis to configure their network and enhance the service quality they offer to their subscribers.

# Dansk Resume

Denne afhandling undersøger performance af pakkekoblede services i dels et isoleret UMTS netværk og hvor UMTS koeksisterer med andre netværk, med fokus på den enkelte link. Både objektive performance mål, såsom nyttekapacitet og pakkeforsinkelse på den enkelte link, og subjektive mål for brugerens oplevelse af servicekvalitet (QoS) indgår i undersøgelsen. Til formålet er der udviklet en realtids emuleringsplatform som benytter en kombination af simulerede og virkelige netværksprotokoller.

For det isolerede UMTS netværk vises der detaljerede emuleringsresultater for performance forbedringer fra flere forskellige mekanismer der kontrollerer UMTS data linklags retransmissioner. Resultaterne viser at det maksimale antal retransmissioner er den mest bestemmende faktor, og at et maksimum på tre og derover eliminerer TCP retransmissioner og fører til optimal performance (op til 85% forøgelse af nyttekapaciteten). Indvirkningen af vilkårlige TCP retransmissioner og forbedringer fra flere anbefalede TCP udvidelser, såsom Eifel algoritmen, er også undersøgt. Eifel algoritmen fører til 33% forbedring i nyttekapaciteten hvorimod DSACK kun forbedrer performance med 14% for "out-of-order delivery" konfigureret i RLC laget. I tilfældet med kortvarige og store pakkeforsinkelser er forbedringerne imidlertid mindre end 9%.

Et andet undersøgt område er indvirkningen fra intersystem handover, og de relaterede buffering teknikker, på de forskellige TCP udvidelser i det tilfælde UMTS koeksisterer med andre netværk. Specielt er der undersøgt en ny teknik benævnt "intermediate buffering". Fuld buffering under handover kan føre til meget lav performance sammenlignet med fravær af buffering, mens "intermediate buffering" giver de bedste resultater. Eifel algoritmen forbedrer performance når handover forsinkelsen er lille og fuld buffering anvendes, men den forringer nyttekapaciteten med op til 34% når fuld buffering ikke

anvendes. Brugen af SACK har ingen indvirkning på fuld buffering og i de tilfælde hvor der forekommer TCP timeouts; dog i tilfældet med handover til større dækning kan der forekomme en væsentlig nedgang i nyttekapaciteten på op til 30%, og modsat en forøgelse på op til 11% for tilfældet med handover til mindre dækning.

I tilføjelse til de objektive undersøgelser er der foretaget eksperimenter omkring såkaldt brugervenlighed for at checke om der er nogen trend i brugerens oplevelse af servicekvalitet, og i så tilfælde hvorvidt de subjektive målinger er i overensstemmelse med de objektive målinger af kvalitet. Resultaterne viser at brugerens oplevelse af kvalitet for det meste er i overensstemmelse med de objektive målinger, hvilket dermed validerer brugen af objektive målinger for undersøgelser af servicekvalitet.

Afhandlingen forsøger at give et holistisk billede af performance for pakkekoblede services ved at sammenholde indvirkningen af forskellige protokollag og ved at tage brugeren i betragtning. Operatører og serviceleverandører kan anvende resultaterne i denne afhandling til konfigurering af deres netværk og dermed forbedre den servicekvalitet de tilbyder deres abonnenter.

# Preface

This PhD. thesis is a result of a three and half years project carried out at the Cellular Systems (CSys) division, now known as the RATE (Radio Access Technologies) Section, in the Department of Electronic Systems, Aalborg University, Denmark. The research work has been conducted in parallel with the mandatory course work, teaching and project work obligations in order to obtain the PhD. Degree. The study has been supervised by Professor Preben E. Mogensen (Aalborg University), Dr. Jeroen Wigard (Nokia Networks) and Associate Professor Troels B. Sørensen (Aalborg University).

The research has been co-financed by Nokia Networks Aalborg R&D and by the Future Adaptive Communication Environment (FACE) and the Center for TeleInFrastruktur (CTIF) Perceived Quality of Service in Heterogeneous networks (POSH) projects at Aalborg University. Specifically, the first half of the project that mainly deals with stand alone UMTS networks is financed by Nokia Networks and the FACE project. The second half of the project that is concerned with UMTS networks in a heterogeneous setting and also user quality perception is co-financed by Nokia Networks and the POSH project.

The thesis is divided into six main parts, namely Introduction (Part I), Background Material (Part II), UMTS (Part III), Heterogeneous networks (Part IV), Subjective Quality (Part V), and Conclusion (Part VI). Part I introduces the thesis and presents the overall objectives and contributions. Part II introduces basic packet service provision concepts in UMTS and related protocols. Part III presents performance results that are concerned with a stand alone UMTS network, while Part IV focuses on results concerning UMTS network performance in a heterogeneous environment. The impact of a network's behaviour on user's perceived quality of service is the subject of Part V. Finally, Part VI gives overall concluding remarks.

Throughout this report citations are in the form of “[reference number]”, where the reference number is an entry in the bibliography which can be found near the end of the thesis. Multiple references are cited in the form “[reference 1, reference 2;..]”. The availability of all online references has been confirmed after the compilation of this thesis. A list of the acronyms used in this thesis is given at the beginning. Five appendices are included at the end to provide supporting information to the main chapters.

# Acknowledgements

This dissertation would not have been possible without the support that I received from many people. First of all, I would like to express my deepest gratitude to my supervisors, not only for their technical guidance and advice but also for believing in me during the most difficult parts of the project. I also appreciate their being patient with me when things were not going as planned. My thanks goes to Troels B. Sørensen, one of my supervisors, for translating the thesis abstract into Danish.

I would like to thank the members of the assessment committee for the interesting discussion during the defence and the detailed comments they provided later on which helped me a lot in the writing of this final thesis.

I would like to thank Lisbeth Schiønning Larsen, for being always willing to help with administrative issues.

Malek Boussif and Kovarththanan Rajaratnam provided me with valuable programming support during the second half of this project, and for this I am greatly indebted to them.

My special thanks goes to Bo Nygaard Bai from the Aalborg University computer workshop for helping me in configuring the network and computers that I used for my experiments.

I am grateful to my colleagues from Aalborg University and Nokia Networks R&D, Aalborg, for participating in the usability experiments that I conducted during the last part of my PhD study. Specially I would like to thank Lars Bo Larsen from the speech and multimedia group for not only participating in the experiments, but also for providing me with books and several ideas regarding usability experiments.

I would also like to thank current and former PhD students; specially Claudio Rosa, Lars T. Berger, Francesco Calabrese, Mohmmad Anas, Akhilesh,

Pokhariyal, Hanane Fathi, and Guillaume Monghal; for making the student life at the university more enjoyable.

Special thanks to my friends Heidi Cruse, Michael Nielsen, Chimeg Uvgun Hanuscheck, Jørgen Peter Hanuscheck, Bruck Sewnet, Amare Gesesse, Samuel Kidane, Akmal Amduka, Mammo Muchie and Fantahun Ketema for the constant support they provided me.

Finally, I would like to thank my family, specially my sisters Kedija and Muna, my mom “Enate” Shemsia, and my brothers Sadik and Siraj, who have always been there for me with love and affection despite the physical distance that separates us.

Oumer M. Teyeb
March 2007

# Acronyms

| | |
|---|---|
| **3G** | Third Generation |
| **3GPP** | Third Generation Partnership Project |
| **AC** | Admission Control |
| **ACK** | Acknowledgement |
| **ACR** | Absolute Category Rating |
| **AM** | Acknowledged Mode |
| **ANOVA** | Analysis of Variance |
| **ARQ** | Automatic Repeat reQuest |
| **awnd** | Advertised Window |
| **BDP** | Bandwidth Delay Product |
| **CCR** | Comparison Category Rating |
| **CDF** | Cumulative Distribution Function |
| **CDMA** | Code Division Multiple Access |
| **C/I** | Carrier to Interference ratio |
| **CN** | Core Network |
| **CRC** | Cyclic Redundancy Check |
| **CS** | Circuit Switched |
| **CTIF** | Center for TeleInFrastruktur |
| **cwnd** | Congestion Window |
| **DCH** | Dedicated Channel |
| **DCR** | Degradation Category Rating |
| **DL** | Down Link |
| **DLL** | Data Link Layer |
| **DS** | Direct Sequence |

| | |
|---|---|
| **DSACK** | Duplicate SACK |
| **E2E** | End to End |
| **EDGE** | Enhanced Data Rates for Global Evolution |
| **EUTRAN** | Evolved UMTS Terrestrial Radio Access Network |
| **FACE** | Future Adaptive Communication Environment |
| **FACH** | Forward Access Channel |
| **FACK** | Forward Acknowledgement |
| **FEC** | Forward Error Correction |
| **FER** | Frame Erasure Rate |
| **FIN** | Finalise |
| **FRTO** | Forward RTO Recovery |
| **FTP** | File Transfer Protocol |
| **GGSN** | Gateway GPRS Support Node |
| **GPRS** | General Packet Radio Service |
| **GSM** | Global System for Mobile communications. |
| **H-ARQ** | Hybrid ARQ |
| **HC** | Handover Control |
| **HSDPA** | High Speed Downlink Packet Access |
| **HSPA** | High Speed Packet Access |
| **HSUPA** | High Speed Uplink Packet Access |
| **IETF** | Internet Engineering Task Force |
| **IP** | Internet Protocol |
| **ITU** | International Telecommunication Union |
| **Kbps** | Kilo bits per second |
| **KHz** | Kilo Hertz |
| **LC** | Load Control |
| **LLC** | Logical Link Control |
| **MAC** | Medium Access Control |
| **Mbps** | Mega bits per second |
| **MMS** | Multimedia Messaging |
| **MSS** | Maximum Segment Size |
| **MTU** | Maximum Transmission Unit |

| | |
|---|---|
| **NACK** | Negative Acknowledgement |
| **NodeB** | Base station |
| **NRT** | Non Real Time |
| **OSI** | Open Systems Interconnection |
| **PC** | Power Control |
| **PDCP** | Packet Data Convergence Protocol |
| **PDU** | Protocol Data Unit |
| **PHY** | Physical |
| **POSH** | Perceived Quality of Service in Heterogeneous networks |
| **PS** | Packet Scheduling |
| **PS** | Packet Switched |
| **QoS** | Quality of Service |
| **RACH** | Random Access Channel |
| **RESPECT** | Real-time Emulator for Service Performance Evalution of Cellular neTworks |
| **RLC** | Radio Link Control |
| **RNC** | Radio Network Controller |
| **RNS** | Radio Network Sub-system |
| **RRC** | Radio Resource Control |
| **RRM** | Radio Resource Management |
| **RT** | Real Time |
| **RTO** | Retransmission Time Out |
| **RTT** | Round Trip Time |
| **RTTVAR** | RTT Variation |
| **SACK** | Selective Acknowledgement |
| **SCC** | Successful Completion Criteria |
| **SCTP** | Stream Control Transmission Protocol |
| **SDU** | Service Data Unit |
| **SIP** | Session Initiation Protocol |
| **SLA** | Service Level Agreement |
| **SMS** | Short Message Service |
| **SN** | Sequence Number |

| | |
|---|---|
| **SNR** | Signal to Noise Ratio |
| **SR** | Selective Repeat |
| **SRTT** | Smoothed RTT |
| **ssthresh** | Slow Start Threshold |
| **SYN** | Synchronisation |
| **TCP** | Transmission Control Protocol |
| **TM** | Transparent Mode |
| **TTI** | Transmission Time Interval |
| **UE** | User Equipment |
| **UL** | Up Link |
| **UM** | Unacknowledged Mode |
| **UMTS** | Universal Mobile Telecommunications System |
| **UTRAN** | UMTS Terrestrial Radio Access Network |
| **VoIP** | Voice over IP |
| **WCDMA** | Wideband CDMA |
| **WLAN** | Wireless Local Area Network |

# Contents

# Part I

# Introduction

# Chapter 1

# Introduction

In this thesis we investigate the quality of packet service provision in Universal Mobile Telecommunications System (UMTS) networks. We will start with introductory material as well as a summary of the overall thesis objectives in this chapter.

The purpose of a data communication network, be it local or wide area, wired or wireless, is to enable the access of different packet services. And different services are characterised by different requirements in terms of a combination of some key performance parameters. The most important parameters are the required data rate, maximum tolerable delay (and its variation, known as delay jitter), and the maximum acceptable information loss ratio [1] (please refer to Appendix A for a definition of some basic networking terms).

The delay and packet loss requirements for some common services is shown in Figure 1.1. As can be seen from the figure, services such as Voice over IP (VoIP) are very sensitive to delays but they tolerate some packet loss. On the other hand, services like File Transfer Protocol (FTP) can not tolerate packet loss at all but are not that sensitive to delay. Different service categorisations exist based on these two basic characteristics (i.e. delay and packet loss), the simplest being Real Time (RT) or Non Real Time (NRT). NRT services can tolerate long delays while RT services are very sensitive to delays. Other categorisations are described in detail in the coming chapters.

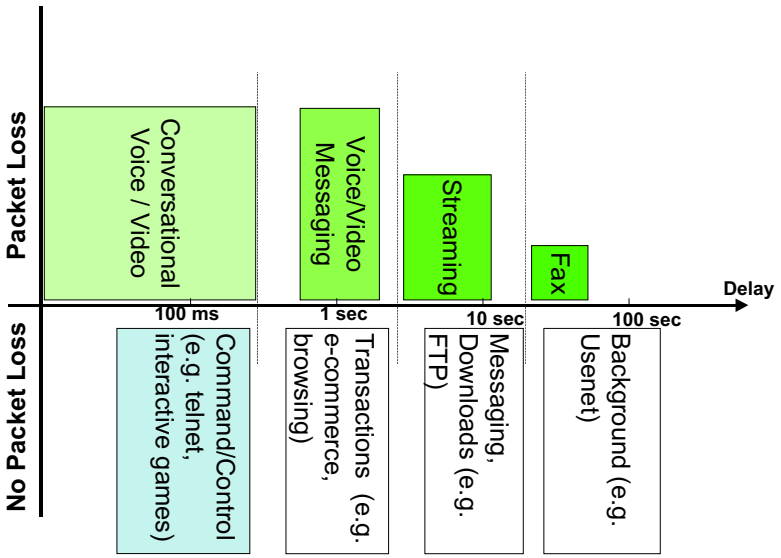



**Figure 1.1:** *Typical delay and packet loss requirements of common services (adapted from [1]).*

The collective effect of service performance that determines the degree of satisfaction of a user of a service is known as Quality of Service (QoS) [2]. It is an umbrella term that is used to cover several aspects of the service such as *availability*, *performance*, *reliability* and *security*. Proper provision of QoS means the proper matching of the network capabilities, the different requirements of several services, and the users (who have different requirements and also different purchasing power). Through proper QoS provision, users will get a guarantee about the services they access, in accordance with what they are willing to pay. Operators and service providers will also be able to optimise their systems to users' need and hence increase their revenue.

## 1.1 Evolution of Radio Access Technologies

The mobile telecommunication industry has grown rapidly during the past few decades, specially after the launch of Global System for Mobile Communications (GSM). GSM enabled voice traffic to go wireless and also provided limited data services such as Short Message Service (SMS) with data rates up to 9.6 Kilo bits per second (Kbps). During the same time, the Internet penetra-

tion rate has grown tremendously, from a mere 0.4% world-wide penetration in 1995 to 14.9% in 2005, and over 50% penetration rate in most European countries [3]. This widespread use of the Internet has propelled the growth of multimedia data services such as web browsing, video streaming, video conferencing and online gaming.

The General Packet Radio Service (GPRS) added a Packet Switched (PS) core network on GSM Circuit Switched (CS) networks, making some of the Internet services such as Multimedia Messaging (MMS) and web browsing accessible via mobile networks. The GPRS air interface offers a peak data rate of 21.4 Kbps per time slot, and multiple time slots can be used by a given user. Theoretically, 8 time slots can be used at once giving a peak data rate of 171 Kbps, but average data rates of GPRS under realistic network load are in the range of 30-40 Kbps [4]. The Round Trip Time (RTT) experienced by packets within a GPRS network is in the order of 500 to 700 ms.

Enhanced Data Rates for Global Evolution (EDGE) is an add-on to GPRS that increases the theoretical peak data rate per time slot up to 59 Kbps (8 time slots per user giving a maximum theoretical peak data rate of 474 Kbps) using new modulation and coding schemes [5]. Under realistic network load conditions, the average data rate offered by the EDGE air interface is around 128 Kbps, and typical RTT values range from 200 to 500 ms [6]. EDGE provides better data rates than GPRS, and hence able to provide more advanced data services such as video telephony. However, like GPRS, it does not support multiplexing of services with different quality requirements over a single connection, e.g. voice, video and packet data.

In order to overcome the limitations of the afore-mentioned systems, and to make the data services that are available on the Internet and furthermore provide other new services, the Third Generation (3G) systems were proposed. 3G networks provide higher data rates of up to 2 Mega bits per second (Mbps) (with average user data rates in the range of 200-300 Kbps), lower RTT values in the range of 150 ms [7] and multiplexing of services with different quality requirements over a single connection [8]. They offer bandwidth on demand, and can satisfy the delay requirements from the delay-sensitive real time traf-

fic to the best effort packet data. The most widespread 3G standard is UMTS, which is based on a Wideband CDMA (WCDMA) air interface. UMTS networks are currently deployed in most of western Europe [9]. The Third Generation Partnership Project (3GPP) is in charge of the standardisation of 3G and its evolutions.

High Speed Downlink Packet Access (HSDPA) is an evolution of UMTS where advanced modulation, coding, and scheduling techniques are used to optimise the Down Link (DL) air interface utilisation based on the channel quality [8]. HSDPA provides lower RTT in the range of 50 ms [10] with theoretical peak data rates of 14 Mbps, and up to 2 Mbps average data rate under practical conditions. High Speed Uplink Packet Access (HSUPA) is the HSDPA equivalent for the Up Link (UL), and it offers theoretical peak data rates of up to 5.8 Mbps, with realistic average values of up to 2 Mbps. The combination of HSDPA and HSUPA is referred to as High Speed Packet Access (HSPA).

With the goal of achieving a very low RTT of about 10 ms [11; 12] and ideal peak data rates of up to 100 Mbps in the DL and 50 Mbps in the UL (corresponding practical values in the range of 2-3 times that of HSPA), the Evolved UMTS Terrestrial Radio Access Network (EUTRAN) standardisation process has been started by the 3GPP [13]. The EUTRAN will be a full-fledged PS network, and currently existing CS services have to be replaced with their PS equivalents (for example, voice services through VoIP).

Figure 1.2 summarises the peak data rate and RTT offered by the mobile communication systems described above.

## 1.2 Motivation

UMTS is one of the first mobile networks that is designed with QoS differentiation in mind, and as such is more flexible towards service provision [8]. The QoS differentiation framework has already been standardised and the deployment of commercial UMTS networks has already started at the beginning

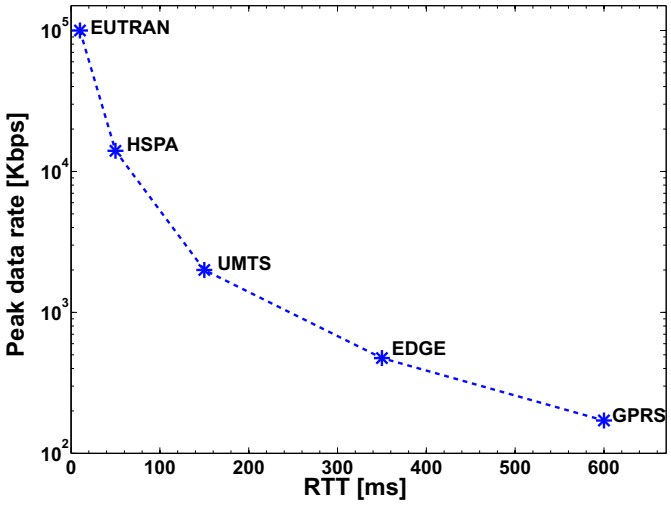



**Figure 1.2:** *Peak DL data rates and average RTT values for different technologies.*

of this PhD study [14]. Due to these reasons, the focus of this project has been put mainly on packet service provision in UMTS rather than the other network technologies described in the previous section.

Proper packet service provision is an interplay of several networking entities and protocols working together to give the user access to services in a transparent way. The task is even more complicated in mobile networks such as UMTS as compared with wired networks due to several factors such as the highly varying characteristics of the wireless channel resulting from interference and mobility. Packet service performance in UMTS is improved by using mechanisms that operate at different layers in the data communication protocol stack (Appendix A).

At the link layer, Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ) mechanisms are used to make the wireless link more reliable [15; 16; 8]. FEC adds redundancy to the packets that are being transmitted so that the receiver is able to correct errors that might occur while the packet is traversing the wireless link. ARQ, on the other hand, requests the retransmission of erroneous packets, in case FEC can not correct the errors. Simulation based studies conducted in [17; 18; 19; 20] have shown the effectiveness of

the ARQ mechanism employed in UMTS. However, previous studies either did not model the link layer protocol of UMTS in detail or they used very simplistic modelling for the higher layer protocols and thus protocol interactions were not properly addressed.

Transmission Control Protocol (TCP) [21] is the most widely used transport protocol in the Internet [22; 23]. Originally designed for wired networks (based on the main assumption that packets are lost only due to network congestion), TCP's reliability and congestion control mechanisms are not optimal for wireless networks [24]. There are several TCP extensions such as TCP Selective Acknowledgement (SACK) [25], Duplicate SACK (DSACK) and Eiffel [26] that try to optimise TCP's performance in a wireless network [27; 28; 29]. Previous studies concerning the evaluation of these different TCP extensions mostly are based on very simplistic link layer models and also they used simulated TCP stacks, which might differ considerably from real world implementations.

UMTS networks are currently (being) deployed in environments where GPRS networks already exist and Wireless Local Area Networks (WLANs) are becoming very accessible via WLAN hot spots in public gathering places such as cafes [30]. Thus, inter-operability issues between UMTS and other technologies is an area of active research [31; 32; 33; 34]. However, most of these studies are concerned with the implementation of different handover mechanisms that try to reduce the handover delay rather than investigate the impact that handover has on quality of packet services.

Most studies concerning packet service performance are based on simulations where the QoS is identified by objective performance metrics such as delay and throughput. However, such studies fail to address the end user who is the final judge of the service quality. And in the few studies where the user's subjective quality metrics are assessed, as in [35; 36; 37], the wireless network is modelled as a simple packet intercepter (delay and/or drop packets).

## 1.3 Objectives

The main objective of this Ph.D. study is:

> ***"To investigate how network protocols (algorithms + parameters), and network conditions affect the quality of non real time packet data services over UMTS Release 99 networks, and to do so from the user's perspective."***

By "user's perspective" it is meant that the study is concerned with the performance of a single connection and that we also address the user's perceived quality.

The objective is addressed through the investigation of:

**Link layer issues:** The Radio Link Control (RLC) layer of UMTS provides a versatile ARQ mechanism and our focus is on optimising its performance for packet services.

**Transport layer issues:** The focus here is to optimise the performance of TCP by investigating TCP extensions proposed to combat performance degradation in wireless networks, taking the UMTS link layer also into account.

**Inter-System handover:** As mentioned previously, inter-operability of UMTS with other networks is an important aspect of packet service performance. As such, the focus here is the optimisation of packet service performance in a heterogeneous network environment in which a user may be handed over from a UMTS network to another network, or vice versa, during a certain service session.

**Subjective quality:** The focus here is the investigation of the impact of the network behaviour on the user's perceived QoS and see if the subjective measures map with objective quality measures.

## 1.4 Assessment Methodology

Evaluation of the performance of different services running over wireless networks can be done in several ways: mathematical analysis, simulation, live testing, emulation or a combination.

Analytical modelling is the most logical step to get some insight on the service performance aspects. However, it turns out to be complex even for a network as simple as a tandem network [38]. Due to this, simplified analytical models are mainly used as the starting point for performing simulation studies.

Simulation offers a simple solution where several scenarios can be investigated in a relatively short amount of time. Studies that focus on lower layer issues usually employ detailed modelling of the wireless channel accompanied with simplified models of higher layers, while the opposite is done for higher layer investigations. However, widely used simulators such as NS-2 [39], usually utilise abstract implementations for the whole network protocol stack, which could differ significantly from real world implementations (e.g. TCP versions used in NS-2 vs. Linux TCP). On top of that, simulation tools usually ignore the effects of operating systems, other concurrently running processes and memory overheads. This could turn out to be the bottleneck rather than the network protocols themselves [40].

Live tests could be performed either on isolated test-beds or even on a real network. Though live testing avoids the problem of using abstract network protocols and entities, the fact that it is almost impossible to change the implementations of the different network protocols or parameter settings just for studying the performance of services limits the scope of this method. Performing live tests can also be very expensive[1]. Results are not reproducible because of the conditions of the live network could vary considerably from one test run to another, which makes it difficult to conduct parameter optimisation tests.

[1] Drive testing, where advanced test phones are driven around the network while measurements are made are traditionally used to access quality measurements by operators. However, such tests are undertaken mostly only as a response to negative customer feedback, due to the expenses incurred [41].

Emulation uses a combination of simulated and real network protocols and components. The parts of the network that are being evaluated are simulated while the other parts are used as in real systems [42; 43]. The advantage of emulation over simulation is that real End to End (E2E) QoS studies can be performed with the aid of end users, and the advantage over live testing is that it is cheaper, and to some extent, it allows the replication of experiments. For these reasons, the results in this thesis are generated through emulations.

The general setup under which the results in this thesis are generated from is shown in Figure 1.3. As can be seen from the figure, two different settings are used during the experiments. In the first case, objective studies are performed without the involvement of the user and in the second case end users are involved to gather subjective results through their ratings.

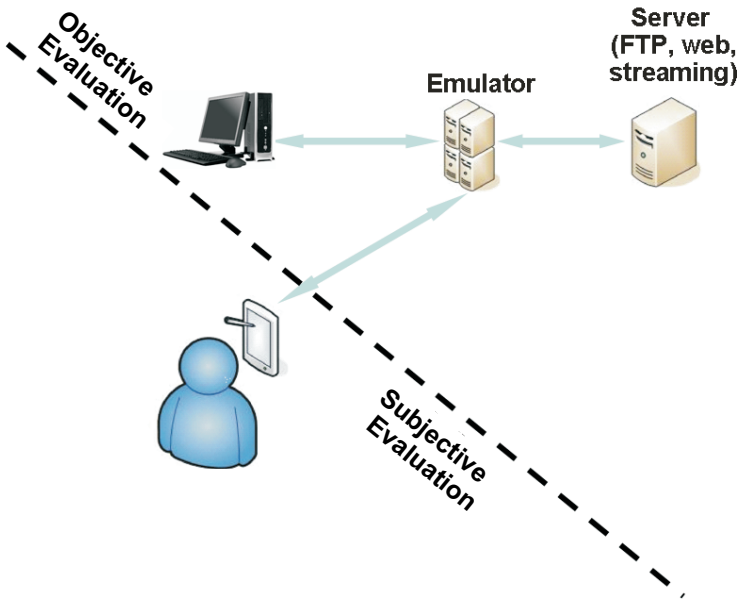



**Figure 1.3:** *Assessment scenario.*

## 1.5 Contributions

This PhD. study started with the design and implementation of a UMTS network emulation platform using which all the investigations in this thesis are performed. The thesis tries to give a holistic view of packet service performance by considering the impact of different protocol layers and taking the end user into account. Operators and service providers can use the results presented in this thesis for enhancing the service quality they offer.

The main contributions of this dissertation are:

- *Optimised UMTS link layer for TCP performance.* The different parameters that govern the UMTS RLC protocol's ARQ mechanism has been optimised for non real time services that employ TCP. The results show that RLC's ARQ is very efficient. The maximum number of retransmissions is the main factor that determines the performance, and setting it to three or more eliminates TCP timeouts and gives optimal results.
- *TCP settings to combat against spurious retransmissions in UMTS.* The new-Reno, SACK, DSACK, Forward Acknowledgement (FACK) and Eifel extensions of TCP has been compared with regard to performance in a UMTS network that is prone to spurious retransmissions caused by packet reordering and delay spikes. Eiffel is found to be very effective against packet reordering but the improvement against delay spikes are not as effective.
- *Enhanced TCP performance in heterogenous networks.* The performance of different TCP extensions in UMTS networks when UMTS coexists with GPRS and WLAN networks has been investigated. A new buffering scheme called intermediate buffering for handover is proposed and its performance is compared with no buffering and full buffering schemes. The results show that intermediate buffering gives the best performance for most of the investigated cases.
- *Understanding of user's perception of packet service quality.* Usability experiments have been carried out for web browsing and streaming ser-

vices in standalone UMTS network and a heterogenous network comprised of UMTS and WLAN. The subjective ratings of the test users were mapped with objective quality measures such as delay and bandwidth. It is found that the different user's ratings mostly agree with each other, and that the subjective rating are in line with the objective measures.

The findings from this PhD. study were presented to the research community in the following articles:

- O. Teyeb, M. Boussif, T.B. Sørensen, J. Wigard and P.E. Mogensen, "RESPECT: A Real-time Emulator for Service Performance Evaluation of Cellular neTworks", in the $62^{th}$ IEEE Vehicular Technology Conference (VTC), September 2005.
- O. Teyeb, M. Boussif, T.B. Sørensen, J. Wigard and P.E. Mogensen, "Emulation Based Performance Investigation of FTP File Downloads over UMTS Dedicated Channels", Lecture Notes in Computer Science (LNCS) 3420, April 2005.
- O. Teyeb, M. Boussif, T.B. Sørensen, J. Wigard and P.E. Mogensen, "The Impact of RLC Delivery Sequence on FTP Performance in UMTS", the $8^{th}$ International Symposium on Wireless Personal Multimedia Communications (WPMC), September 2005.
- O. Teyeb, T.B. Sørensen, J. Wigard and P.E. Mogensen, "Subjective Evaluation of Packet Service Performance in UMTS and Heterogeneous Networks", the $2^{nd}$ ACM International Workshop on QoS and Security for Wireless and Mobile Networks (Q2SWinet), October 2006.

## 1.6 Thesis Outline

The rest of this thesis is organised as follows:

**Chapter 2** presents a general description of packet service provision in UMTS networks. Specifically, the UMTS network architecture, QoS differentiation, and Radio Resource Management (RRM) are discussed. Readers with a strong background in UMTS could skip this chapter without loss of continuity.

**Chapter 3** gives background material on the TCP protocol. The basics of TCP data flow, reliability mechanisms such as retransmissions and congestion control, as well as TCP SACK behaviour are given. This chapter could be skipped by readers familiar with TCP.

**Chapter 4** describes the UMTS RLC protocol. Data transmission within the RLC and the different mechanisms that control RLC's reliability are discussed.

**Chapter 5** evaluates the impact of the RLC protocol on TCP performance. A detailed emulation study along with a simplified analytical model is presented, using which optimal RLC parameter settings are suggested.

**Chapter 6** evaluates the performance of different TCP extensions with regard to spurious retransmissions that occur in a UMTS network. The causes of spurious retransmissions in UMTS networks and TCP extensions that try to mitigate their impact are discussed. The performance of these extensions are evaluated through emulation studies.

**Chapter 7** investigates the performance of packet services in UMTS when UMTS coexists with GPRS and WLAN networks. The performance issue of inter-system handover in heterogeneous networks is discussed. The performance of three buffering schemes on different TCP extensions is evaluated through emulation studies.

**Chapter 8** presents the results from usability tests that were conducted to determine the perceived quality of packet services in UMTS and heterogenous networks. The design of the usability experiments are discussed followed by the analysis of the user ratings and their mapping with objective quality mea-

sures.

**Chapter 9** summarises the main findings from this PhD study and gives an outline of future research topics.

# Part II

# Background Material

# Chapter 2

# Packet Service Provision in UMTS

WCDMA is a wide band Direct Sequence (DS) Code Division Multiple Access (CDMA) system, where user information is spread over a wide bandwidth by multiplying the user data by spreading codes [8]. WCDMA is the air interface of choice for UMTS. The support for variable bit rates, fast power control and soft handover makes it possible for UMTS to support a range of services with different QoS requirements.

Since this thesis is concerned about packet service performance in UMTS Release 99, an understanding of packet service provision in UMTS networks is pivotal to follow the rest of the report. As such, an overview of some of the entities, protocols, and mechanisms that are necessary for proper packet service provision within a UMTS network is given. Section 2.1 gives an overview of UMTS system architecture. The basics of QoS differentiation in UMTS is described in Section 2.2. Section 2.3 is about RRM in UMTS. Finally, the main points in this chapter are summarised in section 2.4. Readers familiar with UMTS can skip this chapter without loss of continuity.

## 2.1 The UMTS System Architecture

A simplified overview of the architecture of a UMTS network is shown in Figure 2.1. The system consists of three major parts: User Equipment (UE), UMTS Terrestrial Radio Access Network (UTRAN) and Core Network (CN). The UE represents the mobile device. The UTRAN handles radio related functions and is comprised of multiple Radio Network Sub-systems (RNSs), each consisting Base stations (NodeBs) and a Radio Network Controller (RNC). The NodeB is the access point to the network, and also provides limited RRM functionality such as fast power control. The RNC, on the other hand, is the main entity responsible for RRM[1]. The CN is responsible for the switching and routing of data connections to external networks and also for keeping databases that are used to manage users.

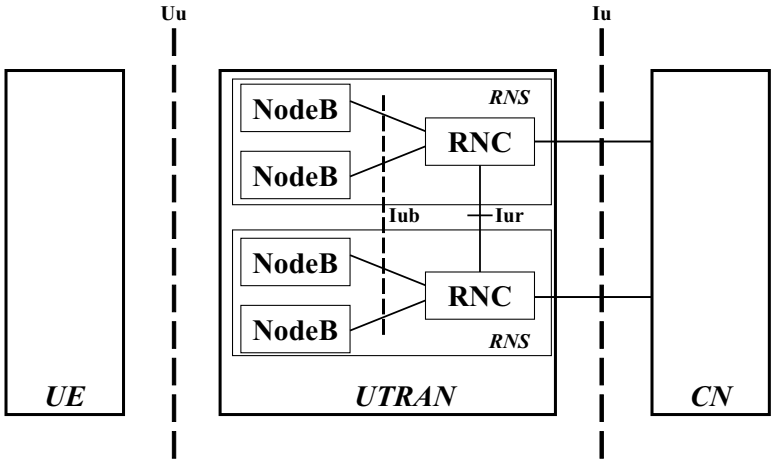



**Figure 2.1:** *Simplified UMTS architecture [8].*

The UE, UTRAN and CN are interconnected via well defined interfaces, and each entity employs protocol(s) with clearly defined responsibilities. Figure 2.2 shows a simplified diagram of the protocols involved in providing a packet service over a UMTS network, and the corresponding entities that terminate these protocols. Note that the CN contains several entities such as Gateway GPRS Support Node (GGSN) that utilise a protocol stack that can go up to the Internet Protocol (IP) layer or higher, but they are not shown here for

[1] In HSDPA, the NodeB takes over most of the RRM functionality of the RNC.

the sake of simplicity.

The UMTS protocol stack is designed with the Open Systems Interconnection (OSI) reference model [44] in mind, and as such each protocol has a well defined responsibility. The most important protocols in the UMTS user plane are the RLC, Medium Access Control (MAC) and Physical (PHY).

The RLC protocol is responsible for the segmentation and transmission of upper layer data. RLC provides retransmission mechanisms that mitigates the losses encountered on the air interface. The details of the RLC retransmission mechanisms are given in Chapter 4.

The MAC layer is responsible for multiplexing/de-multiplexing data from/to upper layers. It is responsible for reporting the UE's buffer occupancy to the RNC for packet scheduling purposes. MAC also controls data flow by notifying the RLC the Protocol Data Unit (PDU) size that it has to use and the number of PDUs that can be sent based on the current selected data rate.

The PHY layer is responsible for transmitting the data over the air interface. It performs operations related with the actual data transmission over the air interface such as FEC coding , Cyclic Redundancy Check (CRC) for error detection, and interleaving to reduce the impact of burst transmission errors.

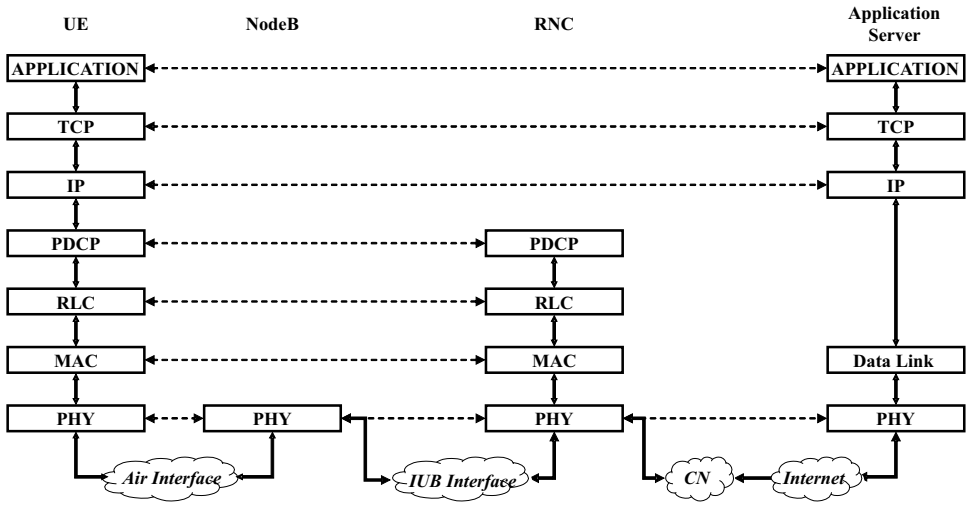



**Figure 2.2:** *Simplified UMTS user plane protocol stack.*

## 2.2 Quality of Service in UMTS

One of the major advantages of UMTS, as compared to 2G systems like GSM, is the extended capability for supporting QoS. QoS is defined as the collective effect of service performances which determine the degree of satisfaction of a user [2]. In order to provide users with a given E2E QoS, UMTS uses a layered bearer service architecture. A *bearer* is an information transmission path of defined capacity, delay, bit error rate etc., and a *bearer service* is a type of telecommunication service that provides the capability of transmission between access points [45]. The UMTS bearer service layered architecture, as shown in Figure 2.3, consists several layers of services with the E2E service being at the top-most layer. This service is realised by the use of bearer services which themselves may use underlying bearer services.

Generally, QoS management is needed because different applications have different traffic characteristics and also differing demands on network performance parameters such as bandwidth, delay and error rate. For example, RT applications are very sensitive to delay, while NRT applications such as file

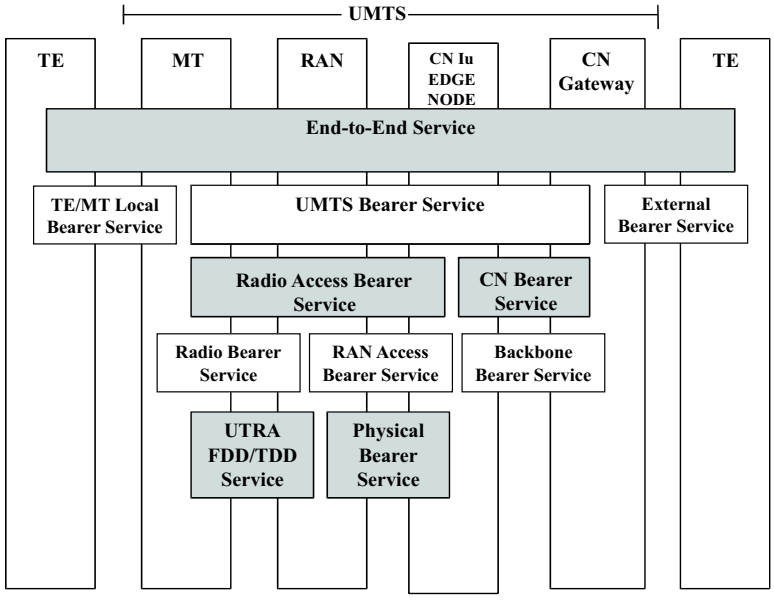



**Figure 2.3:** *UMTS QoS architecture [46].*

transfer are more sensitive to transmission errors. In addition to this, users have different quality standards, while some users may regard high resolution as their main requirement from a video transmission, others may regard a high frame rate as the most important factor. Another important characteristic of the traffic generated by various applications is the traffic symmetry[2].

In 2G systems, such as Global System for Mobile communications (GSM), the provided QoS is fixed and not negotiable. Thus, a delay insensitive application, such as e-mail, would consume the same amount of resources as that of a delay sensitive voice transmission. This results in some applications having more than enough resources allocated, while others having too few, creating an overall inefficient system. UMTS overcomes these deficiencies by grouping the different services under four *QoS classes* (also known as *traffic classes*) [46; 47]:

**Conversational:** The conversational class is characterised by bi-directional and almost symmetric, real-time traffic. This class is the most stringent in terms of delay (max acceptable delay of 400 ms [46; 47]) and jitter (delay variation) requirements, but it is fairly tolerant to packet loss [46]. Services under this category include conversational voice and videophone.

**Streaming:** The streaming class is characterised by an almost real-time and mostly unidirectional DL traffic directed to a human user. Jitter is of the most importance here, but losses can be tolerated as in the conversational class. Streaming is also able to tolerate higher startup delays (upto 10 seconds [47]) as long as jitter is compensated through buffering [8]. Audio and video streaming fall under this class.

**Interactive:** The interactive class carries traffic where the user requests data from a remote server and the response contains the requested data (so there is more traffic in the UL than in the case of the streaming class). Web browsing, voice messaging, and E-mail (server access) fall under

[2] If an application needs more bandwidth for traffic in one direction than in the other direction, the traffic is said to be *asymmetric*.

this category. Jitter is not that much of a problem in this class as compared with the streaming and conversational classes. Delays of upto few seconds are also tolerated [47].

**Background:** If traffic doesn't have precise delay requirements, it is said to be of the background class. This is usually a characteristic of applications where immediate action is not required. E-mail (server-to-server), SMS, and file download are examples of background traffic. The major requirement of this class is that data should be received without error.

Conversational and streaming traffic are collectively known as RT traffic, while interactive and background are referred to as NRT traffic. An application that is running on a UMTS terminal will have to decide which of these QoS classes that it needs as well as handle any renegotiation of the QoS if the need arises[3]. When a request for a certain quality is made by the user, a database in the CN which contains user profiles is checked. If the user's subscription allows the user to access the requested quality, a bearer of the given quality is established. Otherwise, the session has either to be rejected or the user/application might have to lower its standards and require a lower quality bearer in order to access the service.

## 2.3 Radio Resource Management

RRM is the proper utilisation of the radio interface resources, and therefore plays an important role in proper QoS provision in UMTS. The main RRM mechanisms in UMTS are [8]:

**Power Control (PC):** The UL/DL transmission powers are adjusted so that no more power than the minimum necessary to guarantee the required

[3] In addition to the traffic class, additional bearer service attributes can be specified such as guaranteed bit rates, delivery order and transfer delay. The details of these attributes can be found in [46].

Signal to Noise Ratio (SNR) is utilised during transmission, thereby reducing interference. The near-far problem that is prevalent in CDMA based cellular systems is avoided by using a fast power control that operates at a rate of 1.5 Kilo Hertz (KHz), which is fast enough to compensate fast fading on the radio link.

**Handover Control (HC):** By selecting the most optimum cell for a UE to connect with, UMTS's HC reduce interference. UMTS supports *soft handover*, which allows the UE to have multiple active links to several NodeBs during handover, thus avoiding breaks on ongoing sessions[4].

**Admission Control (AC):** This is a procedure performed when a UE wants to start a session. AC checks if the admission of the new call will bring down the quality of ongoing calls or reduce the planned coverage of the system. If so, the new session will be rejected or is given a lower quality than requested.

**Load Control (LC):** When overload situations occur, it is the responsibility of the LC to bring the system back to target load conditions. LC accomplishes this using several methods such as reducing the throughput of packet data traffic, handing over to another carrier or even drop some low priority sessions.

**Packet Scheduling (PS):** PS is responsible for scheduling already admitted NRT calls. Since PS is a very important, if not the most important, aspect of UMTS packet service provision, its details are described in the next sections.

### 2.3.1 Packet Scheduling

The core functionality of the PS in UMTS is located at the RNC, while the NodeB and UE mostly provide measurements that affect the PS functionality.

[4]In *hard handover*, on the other hand, the connection from a certain access point has to be broken before a new connection with another one can be formed. Inter-frequency handover (e.g. between two different cells in GSM) and Inter-system handover (e.g. UMTS to WLAN) require hard handover.

Packet scheduling is done both at the *user* and the *cell* level.

#### 2.3.1.1 User Specific PS

The user specific PS is concerned about allocating the proper transport channel and bit rate for a given user, considering the current traffic volume of the concerned connection. In UMTS, there are two kinds of channels[5]:

**Common Channels:** These are mainly used for signalling but can also carry small amount of user data. The main advantage of common channels is that they do not require any setup time. On the downside, they do not support soft handover (there is a break during cell reselection) and fast power control (not spectrally efficient for sending large amount of data). The common channel in the UL is known as Random Access Channel (RACH), and the DL common channel is called Forward Access Channel (FACH).

**Dedicated Channel (DCH):** These are bidirectional channels, though not necessarily supporting the same data rate in the UL and DL. DCH supports bit rates ranging from a few Kbps up to 384 Kbps. Fast power control and soft handover is supported, but a setup time of around 1 second is required before data can be transmitted on DCH.

Table 2.1 shows RTT traces gathered in January/February 2004 from an operational UMTS network in Aalborg city using the *ping* program[6]. During the process, the UE was moved to see the effects of cell reselection. As can be seen from the table, while using FACH, there is no setup time involved (i.e. no substantial difference between the first two ping times in the traces). However, when cell reselection happens (the bold line in the FACH trace), there is an

[5]There is also a third kind of channel called *Shared Channel* but it is mainly used in HSDPA.

[6]RTT, as defined in Appendix A, refers to the time it takes for a packet to travel from the sender to the receiver and back, and the ping program measures exactly that. However, RTT is also the term used if the receiver is returning some other response than the original received packet.

**Table 2.1:** *Ping traces in FACH and DCH modes.*

| |
|---|
| **FACH** |
| *Pinging trabant.kom.auc.dk [130.225.51.16] with 32 bytes of data:* |
| Reply from 130.225.51.16: bytes=32 time=290 ms TTL=239 |
| Reply from 130.225.51.16: bytes=32 time=270 ms TTL=239 |
| Reply from 130.225.51.16: bytes=32 time=350 ms TTL=239 |
| Reply from 130.225.51.16: bytes=32 time=301 ms TTL=239 |
| ... |
| Reply from 130.225.51.16: bytes=32 time=270 ms TTL=239 |
| **Reply from 130.225.51.16: bytes=32 time=1192 ms TTL=239** |
| Reply from 130.225.51.16: bytes=32 time=371 ms TTL=239 |
| Reply from 130.225.51.16: bytes=32 time=270 ms TTL=239 |
| **DCH** |
| *Pinging www.nokia.com [147.243.3.73] with 128 bytes of data:* |
| **Reply from 147.243.3.73: bytes=128 time=1121 ms TTL=242** |
| Reply from 147.243.3.73: bytes=128 time=221 ms TTL=242 |
| Reply from 147.243.3.73: bytes=128 time=190 ms TTL=242 |
| Reply from 147.243.3.73: bytes=128 time=200 ms TTL=242 |
| Reply from 147.243.3.73: bytes=128 time=190 ms TTL=242 |

increase of around 0.9 seconds, as FACH does not support soft handover (i.e. data can not be transmitted during the cell reselection). On the other hand, from the DCH traces, it can be seen that there is a setup time of 0.9 seconds at the beginning of the trace. Since there is a support for soft handover in DCH, the RTT remains fairly constant even during cell reselection.

A UE can be either in *idle* or *connected* mode. In the idle mode, the UE is able to receive system information and broadcast messages. The Radio Resource Control (RRC) protocol is responsible for establishing the connections between the UE and the UTRAN. The UE enters the connected mode when an RRC connection is established, which is basically a logical connection between the UE and the UTRAN to support data exchange between the higher layers.

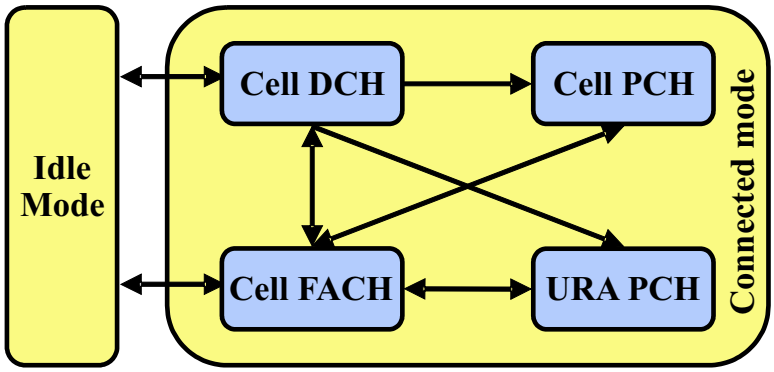



**Figure 2.4:** *UE modes and RRC State Transitions [8].*

Once in connected mode, the RRC can be in different states (see Figure 2.4) depending on the resources available in the network and the amount and frequency of data generated from/to the UE. In *CELL_DCH* state a DCH is allocated for the UE, while in *CELL_FACH* no DCH is allocated, and the UE uses common channels to transmit data. When a long period of inactivity is detected, the state changes to *CELL_PCH* or *URA_PCH*, in which case the UE is accessible only via paging channels and no data transmission is possible. Note that the different states are used in order to utilise network and UE resources efficiently. CELL_PCH and URA_PCH are useful in conserving the UE's battery, while CELL_FACH is useful in managing network resources, as an allocation of DCH reduces the capacity that can be allocated for other users.

The decision whether to be in CELL_DCH or CELL_FACH state depends on the activity level of the UE and whether there are enough resources available to allocate a DCH. Traffic volume measurements are used to decide the activity level. Figure 2.5 gives one example of a possible implementation. In the figure, two threshold values are used: a low threshold to trigger CELL_FACH to CELL_DCH transition and a high threshold to trigger bit rate upgrade. When the buffer occupancy passes the lower threshold with the UE in CELL_FACH, a transition to CELL_DCH is made. When the buffer occupancy increases further and passes the upper threshold, the bit rate is upgraded. Note that we have assumed that there is enough system capacity to allow the CELL_DCH transition and the bit rate upgrade. When the buffer is empty for a

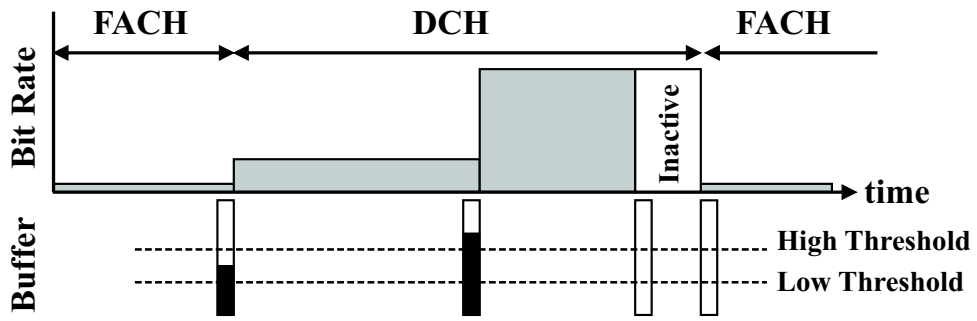



**Figure 2.5:** *Selection between CELL_FACH and CELL_DCH states.*

given period of time, known as *Inactivity Time*, the connection is downgraded to use the CELL_FACH mode again.

Figure 2.6 demonstrates this concept using measurement results from the same network that was used to gather the traces in Table 2.1. Ping RTT measurements are performed continuously on a live UMTS network while increasing the packet size from 1 to 400 bytes, and the channel being used is also logged. From the abrupt reduction in the RTT, it can be concluded that the CELL_FACH to CELL_DCH buffer threshold used in the network is around 250 bytes.

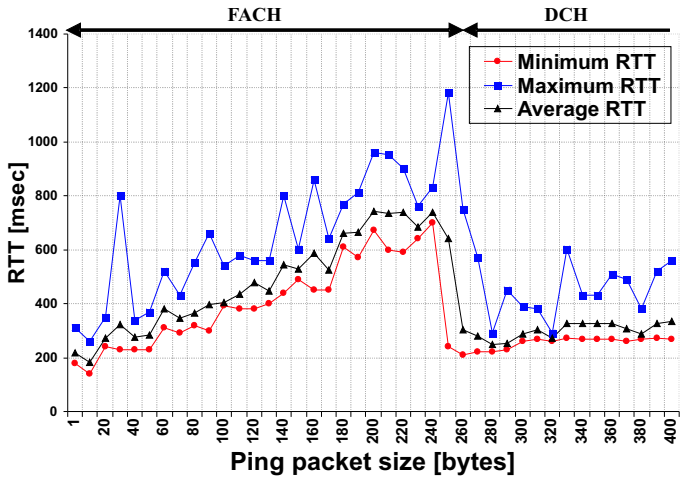



**Figure 2.6:** *Measurement results illustrating CELL_FACH to CELL_DCH transition.*

#### 2.3.1.2 Cell Specific PS

Cell specific PS deals with the sharing of NRT capacity between multiple users, while maintaining the interference levels so that RT traffic is not affected. Basically, the cell specific packet scheduler prioritises the different traffic in the network. Based on the traffic class, *Traffic Handling Priority* (relative importance for handling of Service Data Units (SDUs) of a UMTS bearer compared to that of other bearers) and the *Allocation/Retention Priority* (relative importance compared to other bearers for allocation and retention of UMTS bearer), different traffic class priorities can be defined[7]. Based on these priorities, the bit rate settings, bit rate transitions and ordering of packets of the users (mainly having NRT traffic) will be affected.

On top of the priorities assigned for a specific traffic, different PS algorithms can be used to distribute the system capacity in an efficient way. Although the UMTS PS algorithms have not been standardised, here are some well known algorithms from the open literature [48]:

**Fair Throughput:** Users will have equal throughput wherever they are located, i.e. users with low Carrier to Interference ratio (C/I) will get more resources.

**Fair Resource:** The same amount of power is allocated to the different users and same amount of time is given to each user, i.e. each user gets equal resource.

**C/I based:** Users with a good C/I get more priority than low C/I users. Maximum system capacity is attained at the expense of fairness.

[7] Traffic handling priority is only relevant to interactive traffic, while allocation/retention priority is an optional feature that can be used for all classes. Allocation/retention priority is an issue in admission control while traffic handling priority is relevant during data flow.

## 2.4 Summary

This chapter has provided an overview of packet service provision in UMTS. The UMTS system architecture along with the protocols and mechanisms for providing proper QoS were described.

Release 99 UMTS uses centralised architecture where NodeBs are used for accessing the network and RNCs control the radio resources. Link and physical layer protocols are available that take care of the transmission and reliability details of connections. In order to facilitate the proper provision of QoS, UMTS allows the differentiation of different flows via traffic classes and several other bearer parameters, using which the RNC can schedule the different flows accordingly.

Data transmission in UMTS can be carried using either common channels or dedicated channels. The choice of which channel to use depends on the amount of data to be transmitted, and the available system capacity. While common channels are fast and do not require setup time, they are spectrally inefficient and hence not useful for sending large amount of data. On the other hand, dedicated channels require a considerable setup time but are spectrally efficient, as they support fast power control. Hence, they are most suitable for transmitting larger amount of data.

# Chapter 3

# Transmission Control Protocol

More than two decades of ongoing modifications and optimisations have made TCP the most widespread transport protocol for computer networks. Nowadays, most of the Internet traffic is comprised of TCP flows. TCP averages about 95% of the bytes, 90% of the packets, and 80% of the flows on the Internet [22; 23]. TCP is intended for use as a highly reliable host-to-host protocol between hosts in packet switched computer communication networks [21]. TCP works on top of a non-reliable IP and it provides reliability, flow control and congestion control along with basic data transfer. Though some reliability can be achieved through link layer retransmissions without using reliable transport layer protocols, link layers do not offer congestion control mechanisms, and as such TCP remains the most viable choice for interactive and background services in 3G and beyond networks.

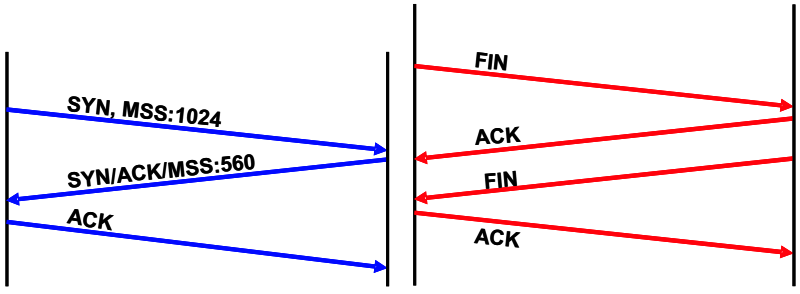
In this chapter, we discuss TCP and some of its extensions. Section 3.1 describes the basics of TCP including connection establishment/termination and data flow. The TCP congestion control mechanisms are described in Section 3.2. Section 3.3 describes the selective acknowledgement extension of TCP. A brief overview of TCP performance in mobile networks is given in Section 3.4. Finally, Section 3.5 summarises the main points from this chapter.

## 3.1 TCP Basics

### 3.1.1 Connection Establishment and Termination

As mentioned earlier, TCP is intended for use as a reliable transport protocol between two hosts. A connection has to be established between the two communicating hosts before data transfer ensues. Connections are also terminated properly at the end of the data transfer, to make the sending and receiving ports available for future connections. A pair of sockets, each specified by a combination of an IP address and port number, uniquely identifies a TCP connection. A TCP packet is composed of a header and a payload. The header is normally 20 bytes long, but can be as large as 60 bytes if all optional fields are used.

TCP connection establishment is a three-way handshake procedure. Figure 3.1(a) illustrates this handshake. The host that is initiating the connection is known as the *client* and the other host is known as the *server*. At the start of a connection the client sends a Synchronisation (SYN) packet, which is simply a TCP header with no payload and the SYN flag set. The Maximum Segment Size (MSS), which is the largest TCP packet, excluding the header, that the client is willing to accept is also specified in the SYN packet. When the server receives the SYN packet, it sends its own SYN packet, with additional information as in the case of the client's SYN. Additionally, it appends an Acknowledgement (ACK) to notify the client that the SYN packet that it has sent has been received properly. The final step in establishing the connection




*(a)*Establishment *(b)*Termination

**Figure 3.1:** *TCP Connection Establishment and Termination.*

is the client's acknowledgement of the SYN packet sent from the server. After this, the connection is ready for data transmission[1].

When the data transmission is done, the connection termination procedure is performed. This is shown in Figure 3.1(b). The host that wants to terminate the connection (which can be either the client or server) sends a Finalise (FIN) packet (a TCP header with the FIN flag set) to its peer host. When the other end receives the FIN, it sends an ACK for the FIN. If this end has no more data to send and also wants to terminate the connection, it sends a FIN packet (after sending the ACK, or along with the ACK, as the ACK and FIN are specified by setting different option fields in the TCP header)[2]. The first host then sends an ACK and after some waiting (to make it possible for the ACK to be resent in case it was lost and the other side has to retransmit the FIN), the connection is closed.

### 3.1.2 TCP Data Flow

Once a connection is established properly, and before the connection is terminated, data can flow between the two TCP hosts. When TCP receives data, it checks the checksum included in the header to see if the data is error free. If the checksum is valid, it sends an ACK[3]. If the checksum is not valid, the packet is discarded. The TCP sender has a timer that it starts when it sends a packet; if the timer expires before it receives an ACK for that packet, the packet is retransmitted. If the receiver gets an out of order packet, it resends the ACK that was sent previously. This is called a *duplicate ACK* and it is used, as will be described later, in detecting losses and triggering retransmissions.

When data flows between two communicating points, a flow control mech-

[1]Common channels can be used for carrying out the connection establishment handshake as the packets involved are small (only TCP headers), but dedicated channels will be typically required for the actual data transmission.

[2]Note that due to the full duplex nature of TCP, a TCP connection can be "half-closed", i.e. one end can close the connection, but the other end could still be able to send more data.

[3]Some TCP implementations delay the ACK for some time. If the receiver has some data to send before this time expires, the ACK is included in that packet. Also, since the ACKs are cumulative, if a new packet is received before this timer expires, the new ACK is sent instead.

anism is necessary to limit the amount of traffic that is passing between them, and as such ensure that the sender will not send data at a rate faster than the receiver can handle. Generally, there are two kinds of flow control mechanisms, *stop-and-wait* and *sliding window* [15].

In *stop-and-wait* flow control, the receiver indicates its willingness to accept data by sending a request to send or an acceptance to receive. The sender then transmits its data. If an ACK is not received within a given time, the packet is retransmitted. If there are several packets to be sent, stop-and-wait will lead to very low link utilisation as the sender will be spending a lot of time just waiting for ACKs.

In *sliding window* flow control, a certain number of packets (called a "window of packets") are transmitted at once, instead of sending just one packet and waiting for its ACK. Each packet has a Sequence Number (SN) and the receiver acknowledges each packet individually, or a certain number of them simultaneously. The sender keeps account of the expected ACK and when it gets it, it advances the window and will be able to send more data. The only time the sender has to be idle is if all the data in a window has been sent but no ACK is received for the first packet in the window. TCP uses the sliding window flow control.

In TCP, the optimal window size is closely related to the capacity of the network in bits. This is known as the Bandwidth Delay Product (BDP), which is the product of bandwidth and RTT [49; 50]. When the window size is equal to the BDP, there will be a constant flow of data from the sender to the receiver, and the link is utilised effectively. If the window size used is smaller than the BDP, the link will be under-utilised while a window that is bigger than the BDP might lead to buffer overflow.

A TCP header contains a field called the Advertised Window (awnd) that is used for reporting the amount of data the receiver is able to handle. Every time a packet is sent, be it an ACK or a data packet, along with it there is the awnd specifying how much data the sender of the packet is willing to accept from the other end. If the receiver's buffer is full when it is sending an ACK, the receiver sets its awnd to zero, which pauses the sender from

sending more data. When more space is available in the receiver's buffer (for example, an application layer has finished reading the buffer), the receiver sends the same ACK as before but with an updated awnd. This ACK is known as *window update*. The sender resumes the data transfer in response to the window update.

## 3.2 Congestion Control and Reliability

Though the sliding window mechanism avoids a fast sender from overwhelming a slow/busy receiver, it is not concerned with the current network congestion level. Hence, the sender can end up using a window size that is appropriate for the receiver but that might cause congestion in the network. TCP uses congestion control mechanisms to make sure the data rate that is used is within the limits of the current network congestion level [49; 51; 52].

### 3.2.1 Slow Start

Slow start is a probing algorithm that is used by TCP to measure the link capacity and network's congestion state. Basically, it means that TCP starts sending small amount of data and starts increasing its sending rate based on the rate of ACK reception. TCP uses a parameter called Congestion Window (cwnd) to control congestion. In the beginning of slow start, the cwnd is set to one[4], and it is incremented by one whenever an ACK acknowledging the reception of a new packet is received.

Actually, the term slow start is a misnomer, as slow start leads to exponential increase in the amount of data that is being sent. This is illustrated in Figure 3.2. The sender uses the *minimum* of the cwnd and the awnd as the window size for the sliding window mechanism. The main drawback to slow start

[4]In [24], initial window sizes of upto four packets is suggested, but this has impact only on the transfer of small data.

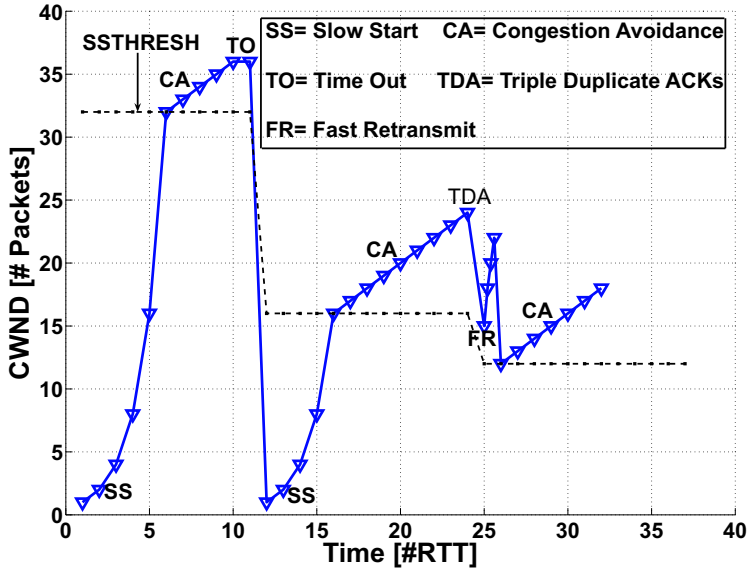



**Figure 3.2:** *TCP Congestion Control.*

is the amount of time that is required during start up before full network utilisation is reached. If the amount of data that is being transmitted is small, or if the RTT is large, the effective bandwidth of the link is reduced considerably.

### 3.2.2 Congestion Avoidance

Congestion avoidance is a congestion control mechanism that is closely interlinked with slow start. Congestion avoidance adds another variable called the Slow Start Threshold (ssthresh). At the start of a TCP connection, cwnd is set to 1, while ssthresh is set to a high value [49]. Whenever an ACK arrives, and if the cwnd is less than ssthresh, the TCP connection is still in slow start phase, and the cwnd is incremented by one. On the other hand, if the cwnd is larger, TCP enters congestion avoidance phase, which makes the cwnd's increment linear instead of exponential. This is usually accomplished by increasing the cwnd by one every RTT instead of every time an ACK is received [52; 49]). Several occurrences of congestion avoidance are shown in Figure 3.2. For example, at the $6^{th}$ RTT, the cwnd becomes equal to the ssthresh and TCP

enters a congestion avoidance phase.

### 3.2.3 Retransmissions

Reliability and congestion control are tightly coupled in TCP because TCP assumes the main reason for packet losses to be congestion and detection of a loss invokes a congestion control procedure. As mentioned earlier, if an ACK for a packet is not received within a certain time, TCP retransmits the packet. The amount of time TCP waits before retransmitting an unacknowledged packet is known as Retransmission Time Out (RTO). The RTO is based on the RTT. Different networks have different RTT, and even in a given connection, due to the variability of the traffic within the network, the queueing delay and the variability of the packet sizes used, the RTT varies from time to time. Therefore, the RTT determination, and hence the RTO should be adaptive. The most widely used RTO estimation is the one that is proposed in [53].

The RTO is initially set to 3 seconds, and TCP uses two intermediate variables known as Smoothed RTT (SRTT) and RTT Variation (RTTVAR) to update it for each RTT sample. When the first RTT sample is gathered, the SRTT is set to the measured RTT, the RTTVAR is set to half of that, and the RTO is set to three times the measured RTT. Afterwards, the variables are updated for each incoming sample ($R(i)$) using the following equations[5]:

$$\begin{aligned} RTTVAR(i) &= \frac{3}{4} \times RTTVAR(i-1) + \\ &\quad \frac{1}{4} \times |SRTT(i-1) - R(i)| \qquad (3.1) \\ SRTT(i) &= \frac{7}{8} \times SRTT(i-1) + \frac{1}{8} \times R(i) \qquad (3.2) \\ RTO(i) &= max(SRTT(i) + 4 \times RTTVAR(i), 1) \qquad (3.3) \end{aligned}$$

[5]In practice, the RTO is usually not allowed to be lower than a certain value. In many operating systems, the lower limit for the RTO is around 1 sec (as shown in Equation 3.3), while Linux allows a 200 ms RTO [54].

The ssthresh is set to half the amount of outstanding data (i.e. number of bytes between the timed out packet and the last transmitted packet) , and the cwnd is set to 1 after a time out, as shown in Figure 3.2 at the $11^{th}$ RTT. This is because TCP assumes a timeout to be a good indication of network congestion, and starts the probing process all over again.

When a timeout occurs, TCP doubles the RTO before retransmitting the unacknowledged packet [53]. If a timeout occurs again, the RTO is doubled again. This continues potentially until the RTO reaches a limit around one minute in most implementations [49], after which it is kept fixed. This technique is known as *exponential back-off* and is essential in keeping the network stable when sudden overload causes packets to be dropped. Eventually, if there is still no successful reception after a certain time, 2-9 minutes depending on the TCP implementations, the connection is reset [49].

### 3.2.4 Fast Retransmission and Fast Recovery

When TCP receives three or more duplicate ACKs in a row (due to packet loss or heavy reordering in the path), it assumes that the concerned packet is lost and it will retransmit it, even if the retransmission timer is below the RTO. This is known as *fast retransmission*. The basic version of TCP that implements the functions of slow start, congestion avoidance and fast retransmission is known as TCP *Tahoe* [55]. In Tahoe, the sender acts the same way after fast retransmission as in the case of a timeout, i.e. slow start after adjusting the ssthresh and cwnd.

A duplicate ACK is an indicator of possible network congestion but it is not as strict as a timeout. If the receiver is sending duplicate ACKs, it is because it is getting packets with higher SN than the missing packet, i.e. there is still some room in the path for data flow. TCP *Reno* is an improvement of Tahoe that utilises this fact [55]. After fast retransmit Reno enters congestion avoidance phase instead of slow start.

Reno also enhances Tahoe by adding *Fast Recovery*. The duration between

the retransmission of a packet due to triple duplicate ACKs and the arrival of a non duplicate ACK signifying the reception of the concerned packet is known as *recovery* period. On entering the fast recovery phase, Reno sets the ssthresh to half of the outstanding data, and cwnd is set to $ssthresh + 3$. The cwnd is increased by three because each duplicate ACK is interpreted as an indication of a packet reaching the destination. For each duplicate ACK that arrives during the recovery period the cwnd is incremented by one, making Reno's cwnd to inflate temporarily. This will make it possible for Reno to send new data while waiting for an ACK for the retransmitted packet. When a non-duplicate ACK arrives (i.e. the missing packet is properly received now), cwnd will be set to ssthresh. In Figure 3.2 at the $25^{th}$ RTT, triple duplicate ACKs were received and the sender enters the recovery phase, where the cwnd, is inflated for a while for each incoming duplicate ACK. When the ACK for the retransmitted packet is received at the $26^{th}$ RTT, the cwnd is deflated back to the ssthresh value and congestion avoidance is resumed.

If multiple packets are dropped, TCP Reno leads to multiple fast retransmits (which means a considerable reduction in the cwnd) or even to a timeout if the multiple losses occur within a window of data [55]. This is because in Reno, the recovery phase is left by the reception of any ACK that acknowledges new data, even though that does not acknowledge all the outstanding packets at the start of the fast recovery. TCP *new-Reno* is an enhancement of TCP Reno that avoids this problem by making sure that the recovery phase is not exited unless all the packets that were outstanding when the recovery phase is entered are ACKed [55; 56].

## 3.3 Selective Acknowledgements

As discussed earlier, the original TCP standard ([21]) and the enhancements Tahoe, Reno and new-Reno use cumulative ACKs, in which only in-sequence delivered packets are acknowledged, and a packet that is delivered out of sequence leads to the generation of a duplicate ACK of the last in-sequence received packet. When multiple packets are lost within a window of data or

when the window size is small, there will not be enough duplicate ACK to trigger fast retransmission, so TCP has to wait for a timeout before retransmitting a packet. On top of that, already sent packets with higher SN than the one that caused the timeout are resent, as the TCP sender is not aware of out of sequence packets that are delivered properly. A retransmission mechanism is referred to as a *Go-back-N* if the loss of packet *N* will lead to the retransmission of every packet with SN greater than N even if they were properly received. TCP's retransmission is not a pure go-back-N as such, since the receiver keeps the out of sequence packets in its buffer and when the lost packet is received, it cumulatively acknowledges all the packets.

With SACK [25], the receiver can inform the sender about all packets that have arrived successfully, whether they are received in or out of sequence. In this way, the sender have to retransmit only the packets that have been lost. SACK uses a TCP option that is comprised of a set of ordered pairs of left and right edge block numbers that specify the sequence numbers of the packets that were received properly. One left-right edge pair takes 8 bytes (4 bytes for one sequence number) and there will be an overhead of 2 bytes for identifying the option is the SACK option. Thus, a maximum of four left-right edge pairs can be put in a TCP header (4*8 +2 = 34), as the maximum size of TCP options is 40 bytes in a given TCP header [6].

The operation of SACK is better illustrated with an example taken from [25]. Suppose 8 packets were sent, each containing 500 bytes, and the first packet has a sequence number of 5000. The second, fourth, sixth and eighth packets were lost. Table 3.1 shows the resulting SACK blocks under this condition. When the third packet is received, there is one out of order received packet, and this leads to the creation of one SACK block signifying this accepted packet, while the cumulative ACK remains the same as before. The situation is similar for the other cases of lost packets, and when the seventh packet is received, there are three SACK blocks signifying the three out of sequence received packets (i.e. packets 3, 5 and 7). Then the second packet was retransmitted (due to three duplicate ACKs reception), but it was delayed

[6] There are some proposals to include more than 4 SACK blocks by using offsets that require less space instead of the full SN [57].

**Table 3.1:** *Example of SACK operation ([25]).*

| Triggering Packet | ACK | First Left Edge | First Right Edge | Second Left Edge | Second Right Edge | Third Left Edge | Third Right Edge |
|---|---|---|---|---|---|---|---|
| 5000 | 5500 | | | | | | |
| 5500 | (lost) | | | | | | |
| 6000 | 5500 | 6000 | 6500 | | | | |
| 6500 | (lost) | | | | | | |
| 7000 | 5500 | 7000 | 7500 | 6000 | 6500 | | |
| 7500 | (lost) | | | | | | |
| 8000 | 5500 | 8000 | 8500 | 7000 | 7500 | 6000 | 6500 |
| 8500 | (lost) | | | | | | |
| 5500 | (delayed) | | | | | | |
| 6500 | 5500 | 6000 | 7500 | 8000 | 8500 | | |
| 5500 | 7500 | 8000 | 8500 | | | | |
| 7500 | 8500 | | | | | | |
| 8500 | 9000 | | | | | | |

in some network element, and the retransmission of the fourth packet reaches the receiver first. At this point, only two SACK blocks are needed because the arrival of the fourth packets has made blocks 2 and 3 to be contiguous. The rest of the table follows a similar pattern.

The congestion control mechanism employed in TCP SACK is similar to new-Reno's, the main difference being selective acknowledgements are used to infer which data should be transmitted [58]. The estimate of the number of outstanding packets is stored in a variable called a *pipe*, and data can be transmitted only when the cwnd is greater than the pipe. When a triple duplicate ACK is received the loss recovery state is entered and the packet indicated by the duplicate ACK is retransmitted. Subsequent retransmission will only be done for those packets that are not SACKed, and the loss recovery phase is not left until all packets that were already sent when the recovery phase was entered are ACKed. In the case of a timeout, SACK information gathered so far is purged and a slow start is initiated [58].

**Forward Acknowledgements** FACK is an extension of SACK where the sender TCP keeps additional variables that keep track of the total number of outstanding data in the network [59]. By using the SN in the SACK blocks

and by counting the retransmitted packets, FACK estimates the total number of outstanding data as:

$$outstanding = snd.next - snd.fack + retran\_data \tag{3.4}$$

where $snd.next$ is the SN of the first packet that is yet to be transmitted for the first time, $snd.fack$ is the SN of the packet next to the forward most acknowledged packet, and $retran\_data$ is the number of outstanding retransmissions.

Data will be sent while the cwnd is greater than the outstanding data. And apart from waiting for three duplicate ACKs and timeouts, retransmissions are triggered also when the reassembly queue is more than 3 packets long, i.e. there are more than three packets between the one referred by the $snd.fack$ and the one referred by the cumulative ACK.

During recovery period (when the receiver is holding non-contiguous data), *cwnd* is kept constant, and when recovery ends, TCP enters congestion avoidance (as described in 3.2.2). Since FACK controls the outstanding data in the network more accurately, it is less bursty than normal TCP or SACK and can recover from episodes of heavy loss in a better way [59].

**Duplicate SACK** In the original SACK specification ([25]), nothing was specified regarding the action to be taken when duplicate packets are received. DSACK is an extension of SACK where the first SACK block is used to report the SN of duplicated packets [60]. A DSACK enabled TCP sender is then able to infer that the duplicate ACK is not due to the reception of a new out of order packet, and hence refrain from mistakenly entering the fast retransmit or fast recovery states. By analysing received DSACK blocks, the sender is also able to distinguish between different sources of duplicate packets such as replication by the network, false retransmit due to reordering, retransmit timeout due to ACK loss, and early retransmit timeout due to a small RTO value.

**Table 3.2:** *DSACK identifying an ACK loss ([60]).*

| Received Packet | Acknowledgement Sent | First SACK block |
|---|---|---|
| 5000 | 5500 (ACK lost) | - |
| 5500 | 6000 (ACK lost) | - |
| 6000 | 6500 (ACK lost) | - |
| 6500 | 7000 (ACK lost) | - |
| timeout | | |
| 5000 | 7000 | 5000-5500 |

An example illustrating the difference between SACK and DSACK is illustrated in Table 3.2, which is taken from [60]. The packets are of 500 bytes long, and the first packet has a sequence number of 5000. As can be seen the first three ACKs were dropped, and the sender times out and retransmits the first packet again. DSACK reports this duplicate reception by including the packet that was duplicated in the first SACK block, even though the cumulative ACK field already included the duplicate packet. If no DSACK was used, only the cumulative ACK will be sent. Since there is a DSACK block identifying duplicate packet reception after a timeout, the sender is able to find out that the no data blocks were actually lost. It can use this information, for example, to undo the reduction of the congestion window due to slow start.

## 3.4 TCP Performance in Mobile Networks

As described in the previous sections, the main features of TCP are its well-designed flow and congestion control mechanisms, which operate on top of its provision of reliability. However, these TCP mechanisms were designed under the assumption that packet losses are only due to network congestion. This assumption holds for wired networks, as packet losses are mainly attributed to buffer overflows caused by congestion. However, in mobile networks, the wireless links are prone to interference which leads to packet corruption. Also, mobility can cause sudden outages and the need to handover to another access point which might lead to loss of packets.

The losses due to interference and mobility are interpreted as congestion

losses by TCP and TCP tries to mitigate the problem by reducing the congestion window as described previously. Doing so will lead to under utilisation of the link bandwidth resulting in significant degradation in the performance.

Figure 3.3 illustrates the performance degradation of TCP due to packet loss. The result is generated by downloading a 1 Mbyte file several times, and randomly dropping packets at different rates. It can be seen from the figure that the packet drop rate has a major impact, a 10% rate leading to more than a 30-fold increase in the download time.

Several methods have been proposed to improve the performance of TCP over wireless networks. The recommendations usually fall into one of the following three categories (please refer to [27; 28; 29] and the references therein for a description and comparison of some of the well known methods):

- **Link Layer:** Recommendations under this category try to make the link layer more reliable, and thereby reduce packet corruption, through error recovery mechanisms such as FEC and ARQ.

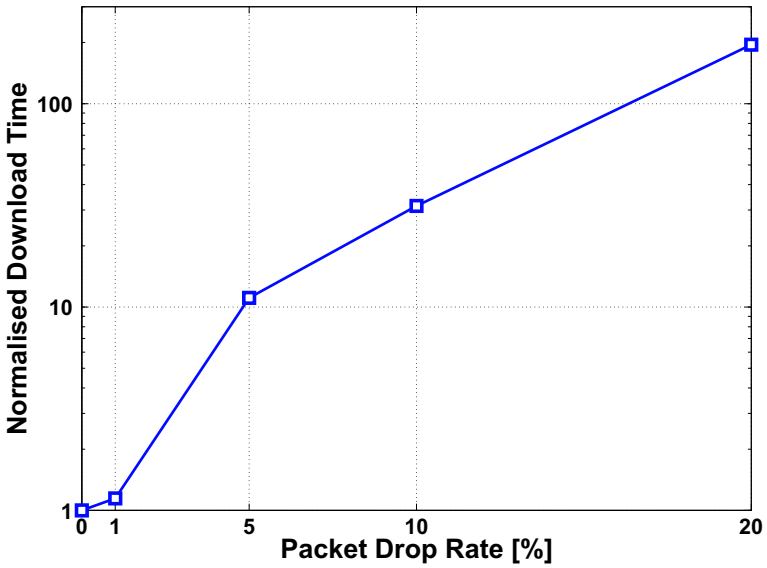



**Figure 3.3:** *TCP performance in the presence of packet errors.*

- **End-to-End:** The recommendation under this category are concerned with modifying the TCP implementation at the sender and/or receiver and/or intermediate routers, or optimising the parameters used by the TCP connection.

- **Split Connection:** The methods under this category try to separate the TCP running on the wireless link from the one that is running on the wired link. The optimisation procedure can then be done separately on the wired and wireless parts.

## 3.5 Summary

During NRT packet data transmission over UMTS, the transport protocol of choice is TCP. TCP is a reliable transport layer protocol. It enforces reliability by acknowledging properly received data and retransmitting unacknowledged data. A TCP connection has to be established before data transmission can ensue and when data transfer is finished, the connection is terminated.

TCP controls the flow of data between the two communicating hosts so that a fast sender will not overwhelm a slow receiver and the overall network will not be congested. TCP does so by using the mechanisms of slow start and congestion avoidance. During the initial stages of data flow, small data will be sent, and for each acknowledgement that the sender gets, the amount of data to be sent is increased by one packet. This is known as slow start. When the sender's output window reaches a certain threshold level, the congestion avoidance phase is entered and the exponential increase is slowed to a linear one.

When a timeout occurs before the reception of an acknowledgement for a packet, TCP retransmits the packet and the sender's window is reduced as TCP assumes a timeout to be an indication of network congestion. When multiple duplicate ACKs are received, TCP retransmits the data instead of waiting for a timeout. By oscillating between higher and lower congestion window size, TCP is able to maintain the congestion in the communication

link at an acceptable level.

Several extensions of the original TCP specification exist, and they mainly try to optimise TCP's behaviour during congestion/packet loss recovery. Tahoe, Reno, new-Reno and SACK are some of the most notable TCP versions.

In wireless networks, TCP's mistaken inference of packet drops due to link errors as congestion losses and the subsequent reduction in the congestion window leads to performance degradation. Adding reliability to the link layer, modifying the TCP implementation or even using separate TCP connections in the wired and wireless part of the network are the main recommendations to mitigate this problem.

# Chapter 4

# Radio Link Control Protocol

According to the OSI reference model (Appendix A) the Data Link Layer (DLL), also known as layer 2, is responsible for providing functional and procedural means to transfer data between network entities and to detect and possibly correct errors which may occur in the PHY layer [44]. The DLL is usually divided into two components: the Logical Link Control (LLC) and MAC. The LLC is responsible for the reliable transfer of upper layer data between two directly connected nodes in the network, while the MAC control access to the media at any given time. In UMTS, the LLC is known as the RLC [61].

This chapter is devoted to the description of RLC and its reliability mechanisms. Section 4.1 describes the basics of RLC, mainly focusing on the data transmission aspects. The different RLC reliability mechanisms are described in Section 4.2. Finally Section 4.3 summarises the main points from this chapter.

## 4.1 RLC Basics

The RLC provides segmentation and assembly for variable length upper layer SDUs. The inter-arrival time between data blocks from the MAC to the PHY is known as Transmission Time Interval (TTI). On the sender side, during

each TTI, a certain number of PDUs are sent from the MAC to the PHY. The PDU size is normally set to the one corresponding to the minimum bit rate for the service (i.e. if the minimum bit rate is 32 Kbps, the PDU size will be 32000 bits/sec $\times$ 0.01 sec/TTI = 320 bits), and multiple PDUs could be fit into one TTI if the bit rate is increased during a given session. The number of PDUs can vary from TTI to TTI, as illustrated in Figure 4.1, depending on the available bit rate and the data at hand. On the receiving side, a certain number of PDUs are forwarded to the RLC, in a manner similar to that of the sender's, but in the opposite direction (i.e. from the PHY to the RLC via the MAC).

The RLC can be configured to operate in three different modes:

**Transparent Mode (TM):** In this mode, as the name implies, no protocol overhead is added to upper layer data, and there is a primitive segmentation and reassembly provision which must be negotiated between the sender and receiver during the radio bearer setup. This mode is suitable for CS services like CS-based streaming, where there is a continuous RT flow of data.

**Unacknowledged Mode (UM):** In this mode segmentation and assembly procedures are facilitated by using protocol headers. Additional functions are also provided, such as timer based discard where SDUs which are not transmitted properly within a given time are discarded. UM is suitable for services such as VoIP.

**Acknowledged Mode (AM):** In this mode, ARQ is used to provide reliability. In this report, the focus is mainly on AM and unless otherwise

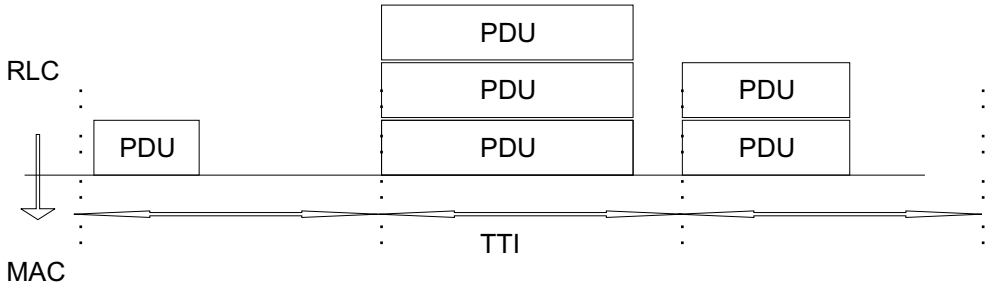



**Figure 4.1:** *Exchange of data between MAC and PHY [62].*

specified, by RLC it is meant RLC operating in AM. RLC's reliability mechanism is described in the latter sections.

During data transmission, the RLC on the sender side receives SDUs from the upper layer. Each SDU is segmented into RLC PDUs or concatenated with others if its size is not big enough to be put into one PDU. Each PDU is given a unique SN. The PDUs are then put in a transmission buffer and scheduled for transmission. Each TTI, the MAC specifies the amount of data that can be sent by the RLC. The RLC complies by sending a given number of PDUs to the MAC, which are then forwarded to the PHY and to the air interface.

On the receiver side, the RLC receives PDUs from the MAC along with their error status, which is checked by the PHY layer using a CRC. The received PDUs are kept in the reception buffer, until all the PDUs that comprise the whole SDU are received, at which point the SDU is assembled. Assembled SDUs are forwarded to upper layers depending on the setting of the parameter *in-sequence delivery*. If in-sequence delivery is set to $false$, assembled SDUs are immediately forwarded to upper layers. On the other hand, if in-sequence delivery is set to $true$, the forwarding depends on the order of arrival of the SDUs, i.e. an SDU that is not assembled sequentially has to wait in the receive buffer till the intermediate SDUs have been assembled and forwarded successfully.

## 4.2 Reliability Provision Mechanisms in the RLC

Depending on the number of different QoS settings during a session, multiple RLC entities can coexist (e.g. using voice and web sessions simultaneously). Each RLC entity has a receiving and a sending side. An overview of RLC's reliability mechanism is shown in Figure 4.2. When the RLC entity is sending data, the sender side is responsible for transmitting the data, while the receiver side is responsible for receiving status information regarding the sent PDUs. During data reception, the receiver side is responsible for receiving the sent PDUs and the transmitting side sends status information regarding the

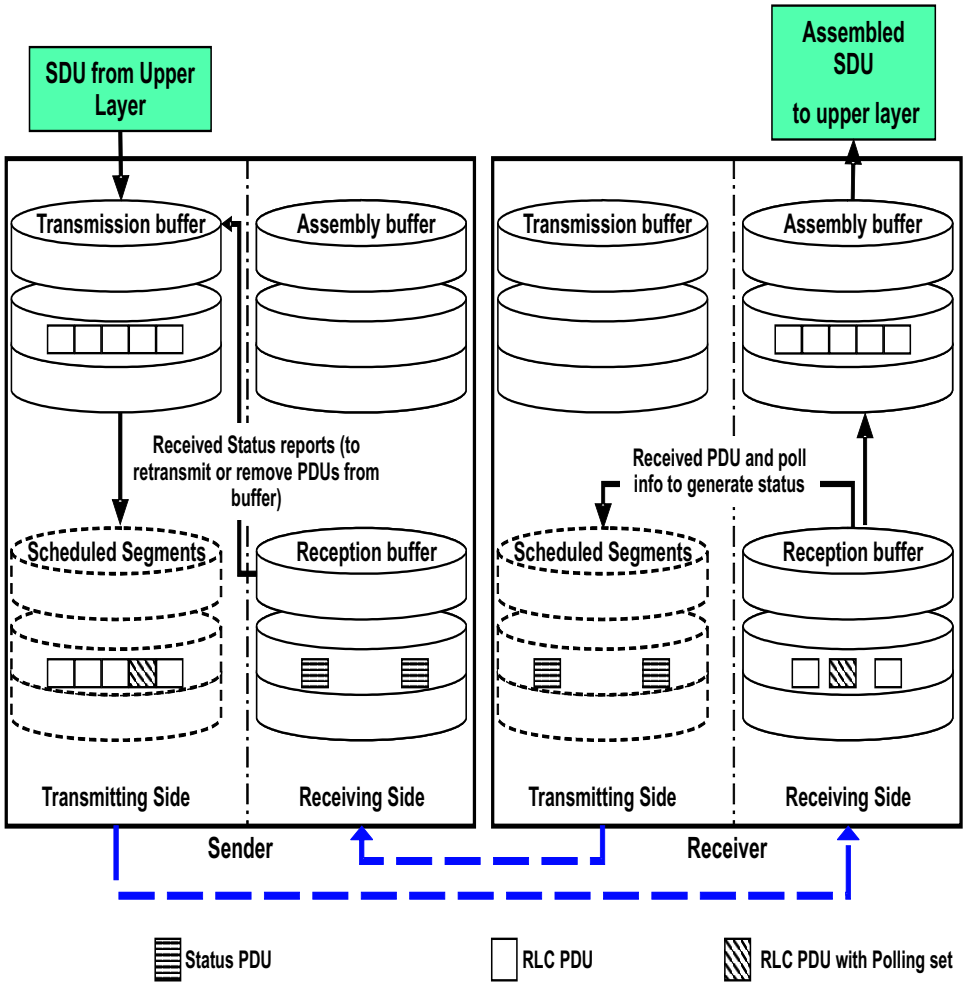



**Figure 4.2:** *A simplified diagram of the RLC data transmission and reliability, where there is only one way flow of data from the sender to the receiver.*

received PDUs. The status information, which is sent via *status PDUs*, is the basis for the RLC's reliability mechanism. The status PDUs can be sent either standalone or concatenated with normal data PDUs.

The status PDUs contain ACKs for the PDUs that are received properly, and Negative Acknowledgements (NACKs) for those that are not. The ACKs and NACKs are encoded in the status PDU using a combination of a cumulative ACK and a bitmap field. A value of $n$ in the cumulative ACK field signifies the correct reception of all PDUs with $SN \leq n$. NACKs and non-cumulative ACKs are sent using the bitmap. For example, if the receiver has received

PDUs #$1, 2, 4, 6$ but not PDUs #3 and 5, it will put 2 in the cumulative ACK field and the values $[0, 1, 0, 1]$ in the bitmap field of the status PDU, where a 0 refers to a NACK and a 1 refers to a non-cumulative ACK. Status reporting is triggered by the sender, the receiver or both.

### 4.2.1 Sender Triggered Status Reporting: Polling

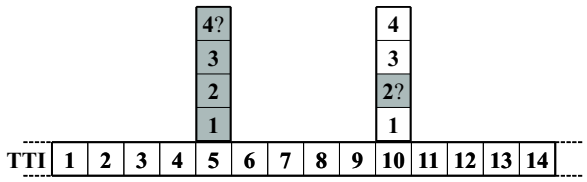
The sender triggers status reporting by setting a *polling* bit in some of the PDUs that it sends. Examples demonstrating the different polling mechanisms in RLC are shown in Figure 4.3. The details of the different polling mechanisms cases illustrated in the figure is given below.

**Poll last PDU and Poll last retransmitted PDU** If *Poll last PDU* is set, the last PDU in the transmission buffer or window that is sent for the first time will be polled. In Figure 4.3(a), PDU #4 is the last PDU in the buffer and it is polled when it is sent at TTI #5. *Poll last retransmitted PDU* is similar to poll last PDU, except that it applies only to retransmitted PDUs. At TTI #10, the sender has received an ACK for all PDUs except for #2, and when this PDU is retransmitted, its poll bit is set.

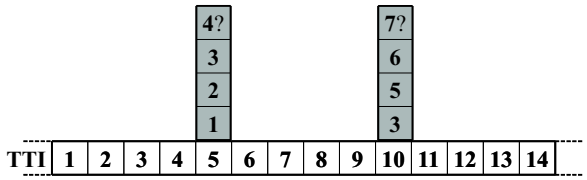
**Poll every poll_PDU** When this is configured, the polling bit is set on every *poll_PDU*$^{th}$ PDU. For example, in Figure 4.3(b), the poll_PDU value is set to 4, and at TTI #5, when the $4^{th}$ PDU is sent, the polling bit is set. At TTI #10, the sender has received an ACK for PDUs #$1, 2$ and 4, and a NACK for #3. When #7 is sent along with the retransmission of #3, the polling bit is set, because it has been four PDUs (#3,5,6, and 7) since the last poll was sent.

**Poll every poll_SDU** When this is configured, the polling bit is set on the last PDU of every *poll_SDU*$^{th}$ SDU. For example, in Figure 4.3(c), poll_SDU is set to 1. Two SDUs were received, the first one which is segmented into two PDUs, and the second one into four PDUs. At TTI #5, when #2 (last PDU of first SDU) is sent, and at TTI#10, when #6 (last PDU of second SDU) is sent, the polling bit is also set.

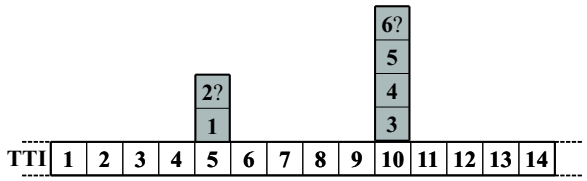
**Window based polling** With *window based polling*, a poll is sent when the percentage of the occupied transmission window reaches a certain threshold. In Figure 4.3(d), this threshold is set to 60%, and the transmission window



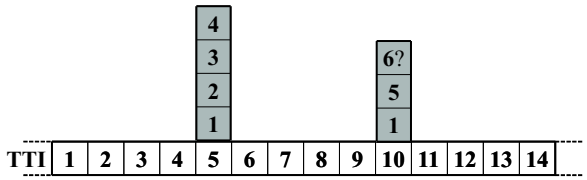

*(a) Poll last PDU and Poll last retransmitted PDU.*



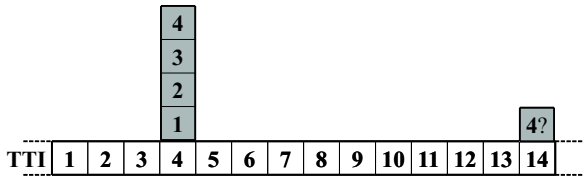

*(b) Poll every poll_PDU.*



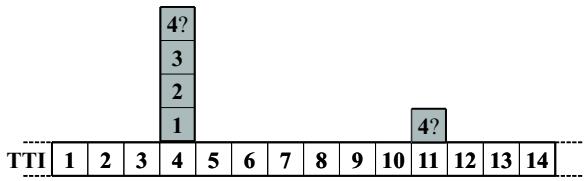

*(c) Poll every poll SDU.*




*(d) Window based polling.*




*(e) Periodic polling.*




*(f) Timer based polling.*


**Figure 4.3:** *RLC polling mechanisms. The columns represents the state of the sender's window at the time instant indicated by the TTI in the horizontal row. With the exception of cases (e) and (f), the temporal spacing between the two instants considered is chosen randomly.*

size is set to 10 PDUs. At TTI #10, even though there are only three PDUs (#1, #5, and 6) that are actually occupying the space in the transmission buffer, the occupied window is considered to be the spacing between the first and last non-ACKed PDUs, which will make the percentage occupied window reach 60%, which triggers the sending of a poll along with PDU #6.

**Periodic polling** If this is configured, a poll is sent regularly at a specified period. In Figure 4.3(e), the period is set to ten TTIs. When the ten TTIs have elapsed since the start of the transmission, which is TTI #14 in this case, a poll is triggered. But during this time, there are no PDUs scheduled to be sent, and as a poll can not be sent without a PDU, one of the PDUs that have not been acknowledged yet will be resent.

**Timer based polling** If this is configured, a timer is started whenever a poll is sent. When the timer expires, if the sender has not received a NACK or a cumulative ACK for the PDU with the highest SN that was waiting for an ACK when the poll was sent, polling will be triggered. In Figure 4.3(f), a poll was sent at TTI #4 and hence the poll timer, which was set to be 7 TTIs, was started too. When this poll was sent, the last PDU that was expecting an ACK was #4. The poll timer expires at TTI #11, and by then neither a NACK nor a cumulative ACK for #4 has been received so a poll will be sent along with the retransmission of #4.

### 4.2.2 Receiver Triggered Status Reporting

Status reporting in the receiver side is triggered either due to the reception of a poll from the sender or other two mechanisms in the receiver RLC. Figure 4.4 illustrates these mechanisms. In Figure 4.4(a), the receiver gets a status request (i.e. a poll) in PDU #4 at TTI #*i*. During the next TTI, the receiver sends a status report. In this case, as everything up to PDU #4 is received, the status contains ACK 4.

The other two mechanisms in RLC that enable the receiver to send a status report without being explicitly polled by the sender are:

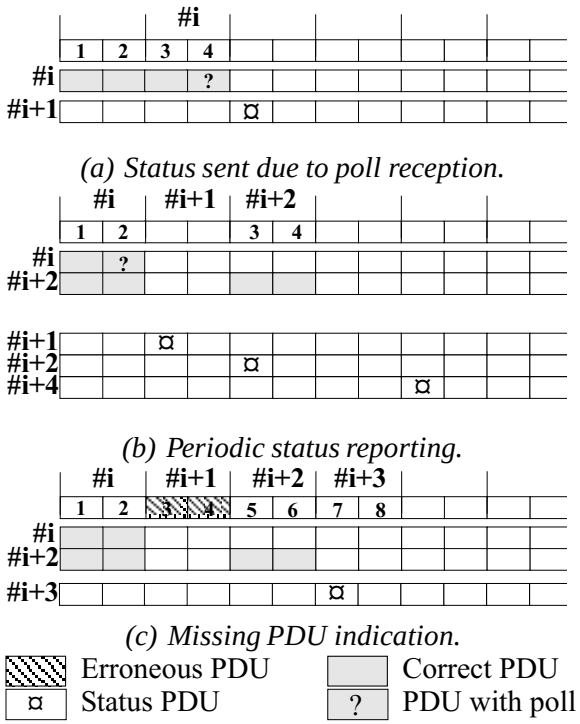



**Figure 4.4:** *RLC status reporting mechanisms. In the sub-figures, the* $1^{st}$ *row represents the PDUs that are being received during the indicated TTI, the* $2^{nd}$ *and* $3^{rd}$ *(set) of rows represent the states of the reception and transmission buffers, respectively, at the receiver side at different TTI instances.*

**Periodic status reporting** When this is configured, a status is sent regularly at a specified period. In figure 4.4(b), the status period is set to two TTIs. At TTI#$i$ a poll is received so the receiver sends a status report (containing ACK 2) at TTI#$i+1$. At TTI #$i+2$, the periodic timer fires for the first time, and since no new PDU has been received till the start of TTI #$i+2$, the same status report is resent. At the end of TTI #$i+2$, PDUs #3 and #4 are received, but as no polling is set on them, no status is sent immediately. At TTI #$i+4$, the periodic status timer fires again and a new status report, containing cumulative ACK 4, is sent.

**Missing PDU indication** If this option is set, a status is sent when the receiver gets PDUs out of order, which is a likely indication that some PDUs may have been lost. In figure 4.4(c), PDUs #3 and #4 are lost in the air interface. The receiver gets PDU #5 while it was expecting #3 at TTI #$i+2$, indicating a missing PDU. During the next TTI, the receiver responds by sending a status

PDU, containing cumulative ACK 2, and a $[0, 0, 1, 1]$ bitmap.

The minimum temporal spacing between consecutive polls and status reports can be controlled by using *Poll Prohibit* and *Status Prohibit* timers, respectively. When these are configured, a timer is started whenever a poll/(status report) is sent, and other polling/(status reporting) triggers that fired while these timers are active will be deferred.

The number of times that a given PDU can be retransmitted can be specified using the *maxDAT* parameter. If a PDU has been retransmitted *maxDAT-1* number of times and still has not been received properly, the SDU that this PDU is part of will be discarded. The sender notifies the receiver when discarding SDUs by sending a status report that contain Move Receiving Window (MRW) indications, so that the receiver will not stall while waiting for parts of an SDU that are not going to come. SDU discard operation can also be specified by setting a timeout value, which is started when the SDU is received from upper layers. If the SDU has not been properly transmitted (i.e. all of its PDUs have not been ACKed) by the time this timer expires, it will be discarded. In services where the maximum delay is an important aspect, this may come handy, as it controls the maximum delay experienced by an SDU.

The different RLC polling and status reporting mechanisms are independent, i.e. they can be enabled at the same time without affecting their individual operation. Using a combination of these mechanisms wisely enables us to avoid deadlock situations that might arise if only one of them is used[1].

## 4.3 Summary

In UMTS, the RLC provides a reliable link layer data transmission by providing ARQ mechanisms. The RLC receives SDUs from upper layers, segments them into RLC PDUs, schedules the PDUs for transmission and then stores

[1] A *deadlock* is a situation that makes the protocol to stop sending data. A typical example of a deadlock is when a sender is waiting for a status report in order to continue its transmission while the receiver is at the same time waiting for new data to trigger a status report.

them in its (re) transmission buffer. Each TTI, the sender transmits a given number of PDUs depending on the instantaneous allocated bit rate on the air interface. The receiver uses status messages to send ACKs for the correctly received PDUs and NACKs for the ones that are not received properly.

There are mechanisms that enable both the sender and the receiver to control the status reporting process, and hence the RLC retransmission. On the sender side, a poll bit can be set in some of the PDUs that are sent to indicate to the receiver that a status report is needed. This polling can be sent either periodically or using some PDU/SDU counters. When the receiver gets these polling requests it complies by sending a status message reflecting the status of its receive buffer. On the other hand, the receiver can control the status reporting by using periodic timers or instantaneously when out of order PDUs are received. In order not to send the polling and status requests at a rate higher than required, there are timers both at the sender and receiver that control the temporal spacing between two consecutive polling requests and status reports, respectively.

# Part III

# UMTS

# Chapter 5

# Packet Service Performance in UMTS: Impact of RLC Parameters

Provision of packet data services is the main driving force behind the standardisation and deployment of 3G networks. The widespread use of TCP makes it the transport protocol of choice for providing these services. As described in Chapter 3, the performance of TCP degrades significantly in mobile networks prone to link errors, and there are several mechanisms to mitigate this problem. The UMTS RLC protocol described in the previous chapter provides one such mechanism, namely link layer retransmission, that can improve TCP performance in the presence of link errors.

In this chapter, the impact of RLC's reliability mechanism settings on FTP services is investigated. A brief survey of related work is given in Section 5.1. Section 5.2 gives a simplified analytical model for evaluating file download times within a UMTS network. Emulation based performance results are given in Section 5.3. Finally, Section 5.4 summarises the main findings of this chapter.

## 5.1 Related Work

The investigation of link layer ARQ performance is an active research area, and it can be classified into two broad categories. The first category includes those that focus on basic ARQ techniques, with the main goal of setting boundaries on the expected delay of an SDU. The research in this category relies heavily on mathematical modelling. The second category contains those studies that focus on the performance of specific systems that employ link layer ARQ with higher layer reliability mechanisms, and is usually done using simulations. A brief survey of related research work is given below. References [63] and [64] fall within the first category, while the rest belong to the second category.

A delay analysis of Selective Repeat (SR) ARQ is given in [63]. Exact and approximate expressions are given for the delay experienced by an SDU in an ARQ channel assuming periodic polling and retransmission timeouts at the link level. The relationship between the delay and SDU size is shown to be quadratic for smaller SDUs and linear for larger SDUs. Also correlated errors are shown to reduce the SDU delay if the error burst length is less than the RTT.

In [64], a method of estimating the average delay experienced by an IP packet in a cellular network with a link layer that supports SR ARQ is presented. Given some statistical parameters such as the radio channel error rate, the IP packet size, and average retransmission times, the model is able to estimate the delay. The results are similar to that presented in [63], the main exception being that the model presented here requires an estimate of a radio block's retransmission delay as an external input parameter.

In [17], a performance study of TCP over UMTS is given. Under the assumption that the link layer is operating a simple go-back-N ARQ mechanism, closed form analytic expression is given for the system throughput. It is shown that TCP performance degrades considerably in a wireless environment as compared with a wired environment. Fortunately, even the simple go-back-N ARQ retransmission mechanism is shown to mitigate most of the

problems.

In [65], an analytical model for TCP file transfer over UMTS is given, where emphasis is put on the slow start phase as a considerable percentage of the file download time can be spent in it for small file downloads. It is shown that the FER has a considerable impact on the end to end performance. In [66], the analytical model given in [65] is further developed by considering limitations on the allowed number of retransmissions at the link layer. The results show that by increasing the number of link layer retransmissions the throughput can be increased but doing so increases the RTT.

In [17], a performance study of TCP over UMTS is given. Under the assumption that the link layer is operating a simple go-back-N ARQ mechanism, closed form analytic expression is given for the system throughput. It is shown that TCP performance degrades considerably in a wireless environment as compared with a wired environment. Fortunately, even the simple go-back-N ARQ retransmission mechanism is shown to mitigate most of the problems.

An optimisation of the UMTS link layer using simulation studies is performed in [18]. The effect of maxDAT and transmission window sizes on TCP performance are presented. The results indicate that higher maxDAT is found to be advantageous and the TCP window size should be set as close as the bandwidth delay product for optimal performance.

In [67], the evolution of the RLC buffer size during file download sessions is presented. The buffer occupancy is compared with the progression of the TCP congestion control stages and the effect of RTT on the buffer size requirements are investigated. The RLC buffer occupancy is shown to reflect TCP's slow start and congestion avoidance stages and that smaller RTT leads to higher buffer occupancy.

In [19], the effect of the different RLC polling mechanisms on the packet call throughput in a TCP session are investigated. The results show that the usage of poll last PDU and poll last retransmitted PDU improves the performance of the system significantly.

In [68], the effect of the different RLC polling mechanisms on throughput and delay are investigated. Settings that lead to the sending of polls spaced apart by around the RTT (where the RTT is defined as the round trip time within UMTS for one radio block) are found to be optimal.

In [20], simulations are performed to find out the effect of different RTT and error rates on the throughput of a TCP connection operating on top of UMTS. The results show that the larger the error rate and the RTT, the larger the TCP window size required for optimal performance.

All the investigations carried out above used either a simplified RLC (for the sake of mathematical modelling), or utilised a simplified TCP model for simulation purposes. The study conducted here differs mainly from the ones mentioned above in that the RLC ARQ mechanism is implemented in detail as specified by the 3GPP, and that a real world TCP implementation (as implemented in the Linux 2.4 kernel) is employed to generate the results. Due to these factors, we believe that our results present the performance of link layer retransmissions in UMTS in a more realistic context.

## 5.2 Theoretical Analysis

In this section, a simplified theoretical model is developed that will be used to compute the delay experienced by IP packets due to link layer retransmissions, and its effect on the download time. The discussion here is not by any means exhaustive, due to the inherent complexity of the RLC ARQ mechanism and its interaction with TCP congestion control algorithms. It is meant to clarify some aspects of these interactions and to provide reference results that will be used to verify the emulation results later in this chapter.

First, approximate expression for the average delay experienced by an IP packet within a UMTS network is derived, and this is used to calculate the time taken for FTP ([69]) based file downloads. FTP uses two parallel connections, one for communicating FTP commands between the sender and receiver, and the other to transfer data packets. In this analysis, only the data channel is

considered and it is assumed that the TCP connection establishment procedure (as described in Chapter 3) is already performed. Also, due to the fast power control in UMTS ([8]) we assume a uniform, uncorrelated error in the air interface [70].

### 5.2.1 Average IP Packet Delay

Assume an IP packet of size $Z$ bits arrives at the RLC (i.e. the impact of the Packet Data Convergence Protocol (PDCP), which can perform header compression, is ignored and an RLC SDU is considered to be an IP packet). Also assume that the RLC PDU size is $K$ bits. That is, there will be $\lceil Z/K \rceil$ PDUs for each IP packet, where $\lceil x \rceil$ refers to the smallest integer greater than or equal to $x$. If the air interface bandwidth is $r$ bps, then $N = \lceil Z/(r * TTI) \rceil$ TTIs (or radio blocks) will be required to transmit all the PDUs at least once.

The probability $P(1, j)$ that $j$ blocks will be retransmitted once is given by:

$$P(1, j) = \binom{N}{j} p^j (1-p)^{N-j} \tag{5.1}$$

where $p$ is the Frame Erasure Rate (FER).

By doing the above calculation recursively (because a radio block that is retransmitted $b$ times must have been retransmitted $b-1$ times before the last retransmission), it can be shown that the probability that $j$ blocks will be retransmitted $i$ times is given by (for $i \geq 2$):

$$P(i, j) = p^j (1-p)^{N-j} \sum_{\substack{a_{i-1}=j}}^{N} \sum_{\substack{a_{i-2}=a_{i-1}}}^{N} \cdots \sum_{\substack{a_1=a_2}}^{N} \{ \binom{N}{a_1} \binom{a_1}{a_2} \cdots \binom{a_{i-1}}{j} p^{(\Sigma_{n=1}^{i-1} a_n)} \} \tag{5.2}$$

where $a_{m-1}$ is an index referring to the number of blocks involved in the $m^{th}$ retransmission round.

From (5.2), the probability $P(i)$ of at least $i$ retransmissions is given by calculating the marginal probability as:

$$P(i) = \sum_{j} P(i, j) \tag{5.3}$$

Figure 5.1 shows the $P(i)$ values while transmitting a 1500 byte IP packet under two different conditions, for a 384 and 32 Kbps connection (the corresponding $N$ values being 4 and 38, respectively), with a 10 ms TTI and 10% FER. As can be seen from the figure, $P(i)$ is negligible, for both cases, for retransmission counts larger than 3, and can be safely ignored for the purpose of this discussion because the normal operating FER for UMTS is within this range [8].

The average delay that an SDU experiences can then be expressed as:

$$SDU\_delay = N + \sum_{i=1}^{\infty} \{P(i)R(i)\} \approx N + \sum_{i=1}^{3} \{P(i)R(i)\} \tag{5.4}$$

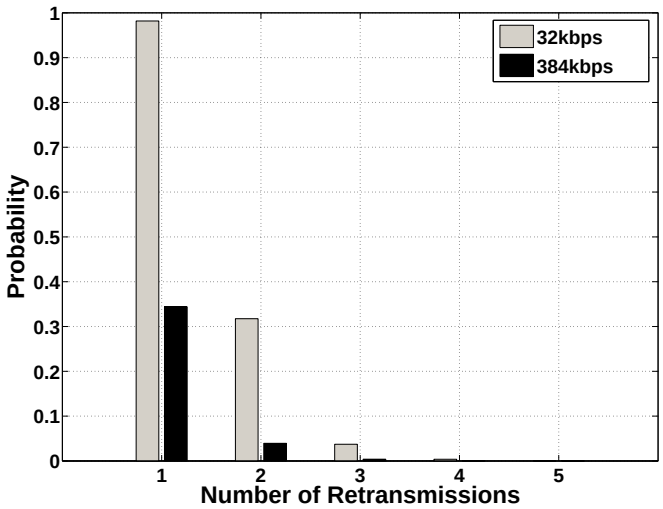
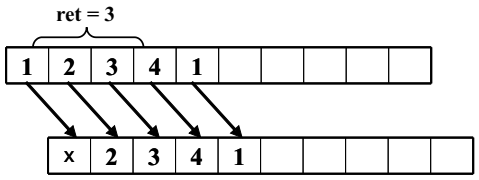
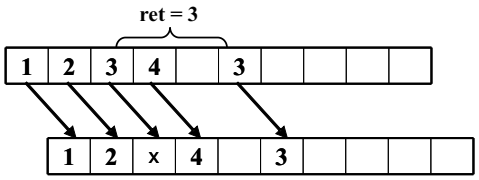


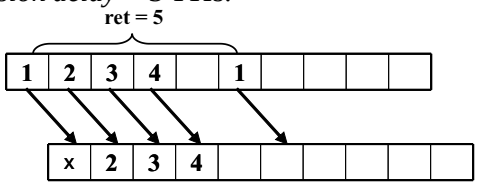

**Figure 5.1:** *The probability of the least number of retransmissions for a 10% FER.*



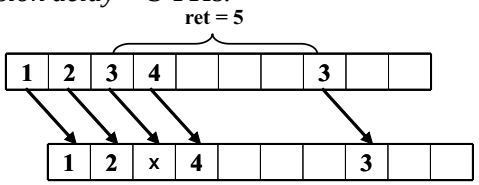

*(a) First block retransmitted, retransmission delay = 3 TTIs.*




*(b) Third block retransmitted, retransmission delay = 3 TTIs.*




*(c) First block retransmitted, retransmission delay = 5 TTIs.*




*(d) Third block retransmitted, retransmission delay = 5 TTIs.*

**Figure 5.2:** *Illustrations of the effect of the retransmitted block position on the delay experienced by an SDU which is comprised of four radio blocks.*

where $R(i)$ is the retransmission delay incurred by the $i^{th}$ retransmission round. $R(i)$ is highly dependent on the polling and status reporting mechanisms used.

For the cases of only one retransmission, the position of the lost block is important. Figure 5.2 illustrates this. If a block at the front is lost (Figure 5.2(a)), its NACK might be received before the rest of the blocks have left the transmission buffer, and hence an extra delay of only one TTI is experienced. If the retransmission delay is larger than the number of blocks (as in Figure 5.2(c)), the extra delay could be more than one TTI, even if it is the first block that is being retransmitted. However, this extra delay is still smaller than the retransmission delay because other radio blocks are sent while the retransmission of the first block is pending. On the other hand, when a block near the back is retransmitted (as in Figures 5.2(b) and 5.2(d)), larger delays are introduced, equalling the maximum value of retransmission delay + 1 if the last block is the one that is being retransmitted. This is because the air interface will be idle for the amount of time equal to the retransmission delay[1], and one more TTI is required to send it via the air interface.

[1]This does not hold when more than one SDU are being processed because blocks representing PDUs from other SDUs can be transmitted during this "idle" period. However, the analysis herein assumes there is only one SDU in the transmission buffer.

If the retransmission delay of one block is *ret* TTIs, then the average delay for the case of only one retransmission round (i.e. no block experiences more than one retransmission) can be calculated as (in terms of TTIs):

$$R(1) = \frac{1}{N}\sum_{j=1}^{N}\{max(1, ret-(N-1-j))\} \tag{5.5}$$

For cases where the retransmission round is greater than one, the probabilities of having more than one block being involved in this round is very small. Figure 5.3 demonstrates this for the case where $N = 10$ and the error rate is 10%. Thus, the extra delay for such cases can be approximated by *ret* (i.e. assuming only one PDU will experience this retransmission round).

Thus the overall SDU delay, i.e. the time from the reception of the SDU at the sender till the correct reception of all the PDUs that belongs to the SDU at the destination, is given by:

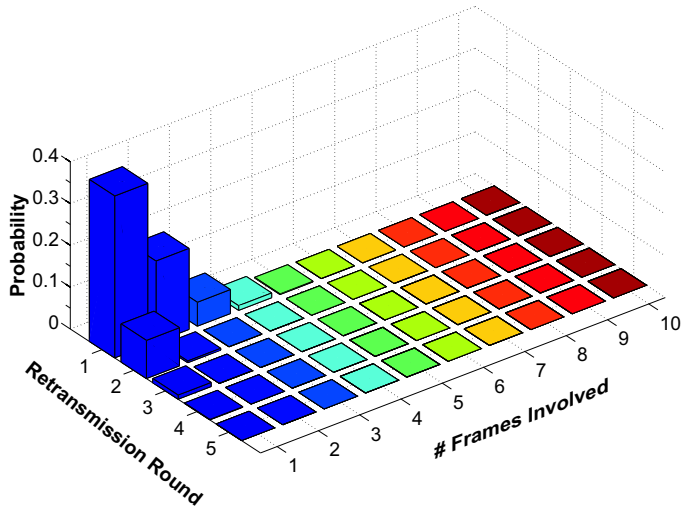



**Figure 5.3:** *The probability of the number of radio blocks involved for each retransmission round, for a 10% FER.*

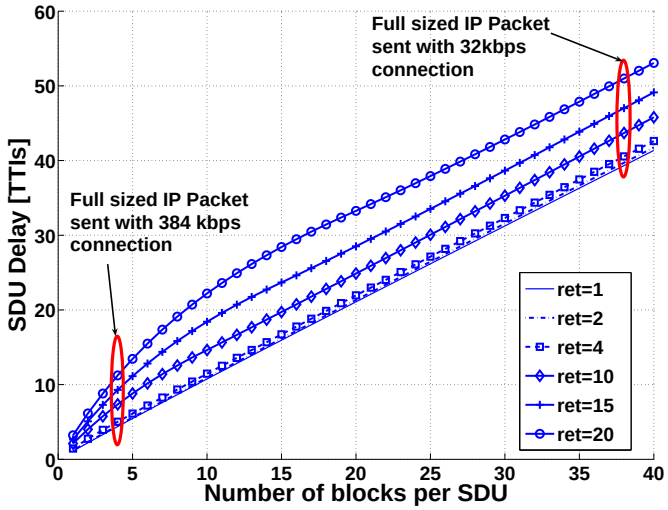



**Figure 5.4:** *Approximate values for the SDU delay for different number of retransmission times.*

$$SDU\_delay \approx N + P(1)R(1) + ret(P(2) + P(3)) \tag{5.6}$$

Figure 5.4 shows the SDU delay for different values of *ret* and number of blocks per SDU, for a 10% FER. The result shows similar trend to the one shown in [63] (as can be seen from the figure, for smaller values of $N$, the relationship between $N$ and SDU delay is quadratic but it becomes linear when $N$ is large), though exact comparison between the two is not possible because of the different assumptions and parameters used in the two cases.

### 5.2.2 File Transfer

In the previous section an approximate value for the average delay experienced by an SDU was calculated. This value is used in this section to determine the average file transfer time.

We assume the connection will always be in slow start, i.e. there are no packet losses due to errors in the air interface thanks to link layer retransmissions and the cwnd will not reach the ssthresh. This is a valid assumption

because in many implementations of TCP (e.g. Linux), the ssthresh is usually set to a very high value, which means as long as a timeout or triple duplicate ACKs do not occur, TCP will be in slow start. Also, for the considered UMTS network, the maximum BDP is 7200 bytes (for 384 Kbps connection with an RTT of 150 ms), which allows only five full size (1460 byte) packets to be outstanding at any given time, i.e. we reach the TCP steady state very quickly, even for files as small as 10 Kbyte. Congestion is not considered in the wired part as the focus here is on the wireless part, and we do not consider congestion in the wireless part as the radio resource management of UMTS has a tighter control of the transmission rate of each connection.

Consider a UE downloading a file from a server over the Internet using FTP, i.e. packets are flowing in the DL direction and only ACKs are transmitted in the UL. The file is of size $filesize$ and the number of packets required to transmit the file is given by $M = \lceil filesize/Z \rceil$, where $Z$ is the average size of one IP segment, i.e. the size of the packet without headers. Figure 5.5 shows a sequence diagram of the data flow, without considering the connection setup and connection termination. The alternating shades in the picture depict different cycles, i.e. periods between the time a packet is sent and the time of arrival of its corresponding ACK.

$u\_dl$ and $u\_ul$ refer to the rate of packet transmission (in $packets/sec$) in the air interface in the DL and UL directions. They are given by:

$$u\_\{ul,dl\} \quad = \quad r\_\{ul,dl\}\frac{(1-FER\_\{ul,dl\})}{(pkt\_\{ul,dl\}))} \tag{5.7}$$

where the $r\_\{ul,dl\}$ and $pkt\_\{ul,dl\}$ refers to the connection bit rate and the average packet size in the corresponding direction, respectively. $pkt\_ul$ is assumed to be 40 bytes, as it contains only a TCP/IP header, and $pkt\_dl$ is assumed to be 1500 bytes, which is a typical packet size for bulk data flows.

$RTT$ is the round trip time of one IP packet and it is given by:

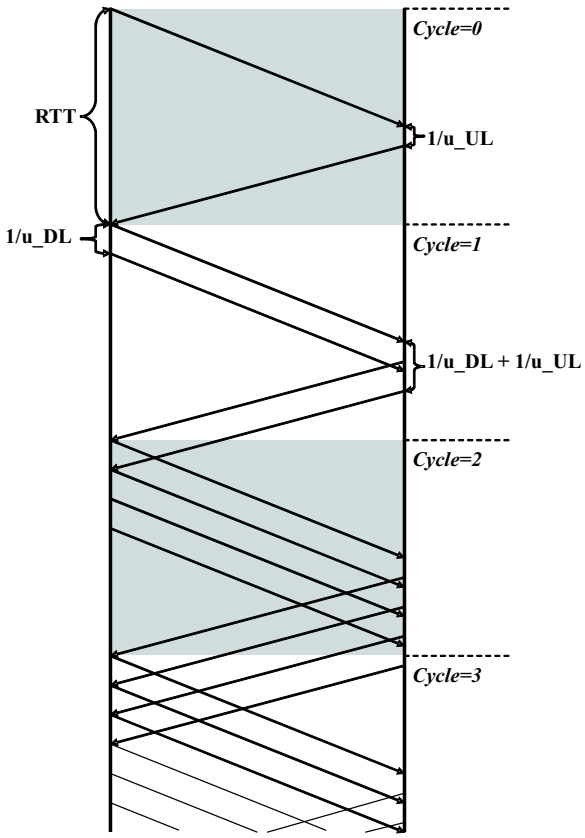



**Figure 5.5:** *A sequence diagram showing the first few cycles during an FTP session.*

$$\begin{aligned} RTT &= SDU\_delay\_dl + SDU\_delay\_ul + \\ &\quad RTT\_UMTS + RTT\_Internet \end{aligned} \tag{5.8}$$

where $SDU\_delay\_dl$ is the SDU delay (as calculated using 5.6) for a packet of size $pkt\_dl$ in the DL direction, and $SDU\_delay\_ul$ is the SDU delay for one packet of size $pkt\_ul$ (i.e. the ACK for the corresponding DL packet) in the UL direction. $RTT\_UMTS$ is the RTT within the UMTS network (i.e. the average processing times, independent of the packet size, in the UTRAN and CN) without considering the air interface transmission time (as this is basically $SDU\_delay\_ul + SDU\_delay\_dl$), and $RTT\_Internet$ corresponds

**Table 5.1:** *Packet departures.*

| Cycle | Packet ID | Start of Transmission | End of Transmission |
|---|---|---|---|
| 0 | 1 | 0 | $\frac{1}{u\_dl}$ |
| 1 | 2 | $RTT$ | $RTT+\frac{1}{u\_dl}$ |
| | 3 | $RTT+\frac{1}{u\_dl}$ | $RTT+\frac{2}{u\_dl}$ |
| 2 | 4 | $2\times RTT$ | $2\times RTT+\frac{1}{u\_dl}$ |
| | 5 | $2\times RTT+\frac{1}{u\_dl}$ | $2\times RTT+\frac{2}{u\_dl}$ |
| | 6 | $2\times RTT+\frac{2}{u\_dl}$ | $2\times RTT+\frac{3}{u\_dl}$ |
| | 7 | $2\times RTT+\frac{3}{u\_dl}$ | $2\times RTT+\frac{4}{u\_dl}$ |
| ... | ... | ... | ... |
| i | $2^i$ | $i\times RTT$ | $i\times RTT+\frac{1}{u\_dl}$ |
| | $2^i+1$ | $i\times RTT+\frac{1}{u\_dl}$ | $i\times RTT+\frac{2}{u\_dl}$ |
| | $2^i+2$ | $i\times RTT+\frac{2}{u\_dl}$ | $i\times RTT+\frac{3}{u\_dl}$ |
| | ... | ... | ... |
| | $2^{i+1}-1$ | $i\times RTT+\frac{2^i-1}{u\_dl}$ | $i\times RTT+\frac{2^i}{u\_dl}$ |
| i+1 | $2^{i+1}$ | $(i+1)\times RTT$ | $(i+1)\times RTT+\frac{1}{u\_dl}$ |
| | ... | ... | ... |

**Table 5.2:** *ACK arrivals.*

| Cycle | ACK received for Packet # | Time | cwnd |
|---|---|---|---|
| 0 | - | - | 1 |
| 1 | 1 | $RTT$ | 2 |
| 2 | 2 | $2\times RTT$ | 3 |
| | 3 | $2\times RTT+\frac{1}{u\_dl}$ | 4 |
| 3 | 4 | $3\times RTT$ | 5 |
| | 5 | $3\times RTT+\frac{1}{u\_dl}$ | 6 |
| | 6 | $3\times RTT+\frac{2}{u\_dl}$ | 7 |
| | 7 | $3\times RTT+\frac{3}{u\_dl}$ | 8 |
| ... | ... | ... | |
| i | $2^{i-1}$ | $i\times RTT$ | $2^{i-1}+1$ |
| | $2^{i-1}+1$ | $i\times RTT+\frac{1}{u\_dl}$ | $2^{i-1}+2$ |
| | $2^{i-1}+2$ | $i\times RTT+\frac{2}{u\_dl}$ | $2^{i-1}+3$ |
| | ... | ... | ... |
| | $2^i-1$ | $i\times RTT+\frac{2^{i-1}-1}{u\_dl}$ | $2^i$ |
| i+1 | $2^i$ | $(i+1)\times RTT$ | $2^i+1$ |
| | ... | ... | ... |

to the RTT contribution from transmission within the Internet.

Table 5.1 shows the packet sending sequence and Table 5.2 shows the time of arrivals of the ACKs which trigger the transmission of the data packets (except for the first packet which was sent after the initial TCP handshake as described in Chapter 3). As can be seen from Figure 5.5 at the beginning

there are some cycles where there is a transmission gap as the sender waits for ACKs. However, this gap decreases as the data transmission continues and a situation will arise where there will be no gap. This happens when the last packet from a given cycle is not sent yet when a new cycle starts. For example, if the ACK for packet #4 is received before packet #7 is sent, it means full link utilisation from then onwards.

Let $C_f$ denote the cycle from which there will be full link utilisation, i.e. the last transmission gap occurs in cycle $C_f - 1$. This means by the time the last packet in cycle $C_f$ is transmitted (which occurs at $C_f \times RTT + 2^{C_f}/u\_dl$ as can be seen from Table 5.1), the ACK for the first packet from cycle $C_f$ is received (which happens at $(C_f + 1) \times RTT$ as can be seen from Table 5.2). Thus, we have the relation:

$$\begin{aligned} C_f \times RTT + \frac{2^{C_f}}{u\_dl} &\geq (C_f + 1) \times RTT \\ \Rightarrow C_f &= \lceil \log_2(RTT \times u\_dl) \rceil \end{aligned} \tag{5.9}$$

the $\lceil\ \rceil$ operator is used, because $C_f$ is the cycle during which this overlap occurs for the first time and the cycle number must be an integer.

The total number of cycles needed to transmit the file, $C_{last}$, is given by (using the fact that by the time cycle $i$ has finished, $2^{i+1} - 1$ packets are sent):

$$\begin{aligned} 2^{C_{last}+1} - 1 &\geq M \\ \Rightarrow C_{last} &= \lceil log_2(M+1) - 1 \rceil \end{aligned} \tag{5.10}$$

The total transmission time can be divided into two periods: before and after full link utilisation. However, if $C_{last} < C_f$, the file download is finished before full link utilisation is reached, and as such the file transmission time is divided into two periods: before the last cycle and during the last cycle. The time spent to reach any cycle $i$ is simply the time spent from 0 to $i - 1$. Thus the first period is given by:

$$T_{first} = min(C_f, C_{last}) * RTT \tag{5.11}$$

From Table 5.1 it can be seen that by the time cycle $i-1$ has finished, $2^i-1$ packets are sent. Thus, the number of packets to be sent during the second period is given by:

$$remaining = M - (2^{min(C_f, C_{last})} - 1) \tag{5.12}$$

The time required to send the remaining packets is:

$$T_{second} = \frac{remaining}{u\_dl} \tag{5.13}$$

Thus, the total time required to send the file becomes (i.e. the time from the transmission of the first packet to the reception of the last packet at the destination):

$$T_{total} = T_{first} + T_{second} \tag{5.14}$$

Figures 5.6, 5.7 and 5.8 show the download time (5.14), the RTT (5.8) and the link utilisation while downloading a 100 Kbyte file over a 128 Kbps connection. Based on the values given in [8] and [7], the $RTT\_UMTS$ is taken to be 115 ms . A value of 50 ms is used for the $RTT\_Internet$ [2]. The retransmission delay has no effect for the error free connection, as expected, and the effect becomes more pronounced as the error rate increases as can be seen from the slopes of the different curves. For the 20% FER case, the average RTT is almost tripled when the retransmission delay is increased from 1 to 30 TTIs. However, this does not directly translate to similar proportional increase in the download time because once the full link utilisation cycle has started (for the considered cases here the maximum value of $C_f$ is 3), the effect of the RTT is minimal due to the continuous flow of data. For small file sizes or/and higher values of $C_f$, the effect of the RTT will be more pronounced as

[2] 50 ms was used as it was found to be the average RTT found through ping from well known sites such as Google from Aalborg University servers.

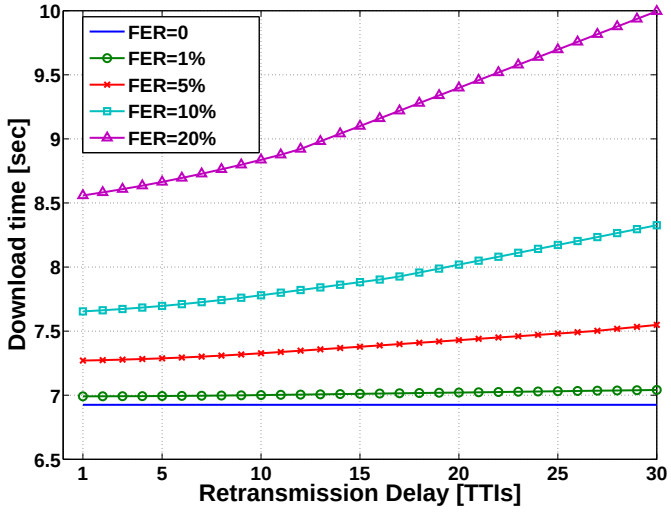



**Figure 5.6:** *Download times of a 100 Kbyte file under different retransmission delays and FERs.*

that means more time is spent in cycles with transmission gaps. For example, if the file size is less than 1 Kbyte, the file download time basically maps to the RTT, as only one packet is necessary to send the whole file.

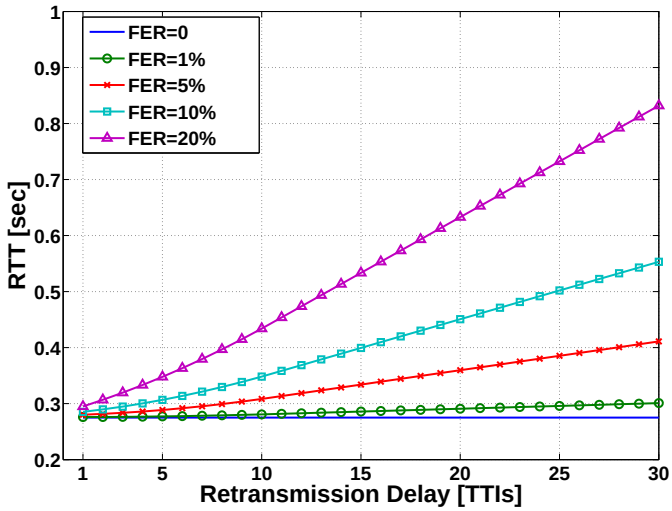



**Figure 5.7:** *RTT during a download of a 100 Kbyte file under different retransmission delays and FERs.*

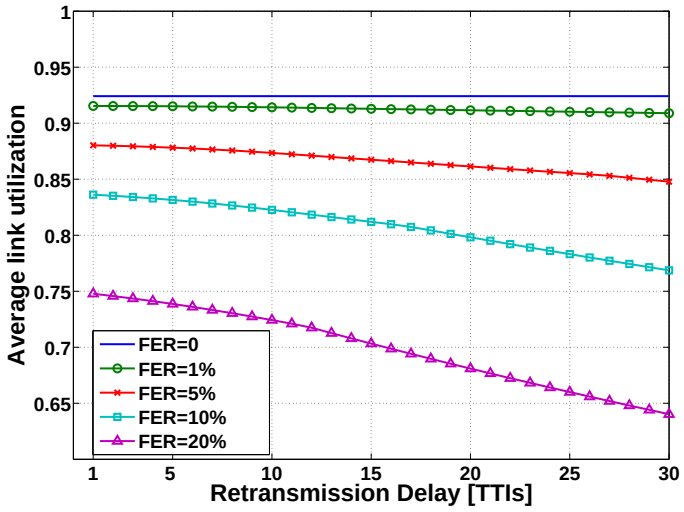



**Figure 5.8:** *Link utilisation during a download of a 100 Kbyte file under different retransmission delays and FERs.*

## 5.3 Emulation Results

As described in section 1.4, the investigations in this thesis are carried out using emulations. To this end, an emulation platform has been developed. A brief description of the emulator, a listing of the main assumptions and parameters, and the emulation results are given in the following sections.

### 5.3.1 Emulation Tool and Parameter Settings

### 5.3.2 Emulation Tool

The studies are carried out using Real-time Emulator for Service Performance Evalution of Cellular neTworks (RESPECT) [71; 72]. RESPECT provides a link-level, real-time emulation of a UMTS network using free software tools that are available with standard Linux installations. The emulator's usage scenario is already described in Chapter 1 (see Figure 1.3). The emulator runs on a PC that is also configured to act as a proxy server, and users are con-

nected to the application server (possibly via the Internet) through this machine. Users specify their emulation setup using configuration files, which includes information such as the data rate, the error rate, processing delays of different network entities, and the different RLC parameters discussed in the previous chapter.

The functionality of the emulator, as seen from a protocol point of view, is depicted in Figure 5.9. The dotted box represents the parts of the protocol stack in the emulator machine that are relevant to us. In a nutshell, RESPECT inserts the additional protocol layers that comprise a UMTS network between the IP and data link layers of a normal TCP/IP protocol stack. In a real UMTS network these layers are distributed in different network entities, as shown by the names in italics, such as the UE, NodeB, and RNC. In RESPECT, they are all lumped together into one entity, and doing so does not have different characteristics from that of the real case, as long as the processing delay of the different network entities are taken into account.

A user starts a network-based application such as a FTP session or video streaming, which is accessed via the proxy server where the emulator is run-

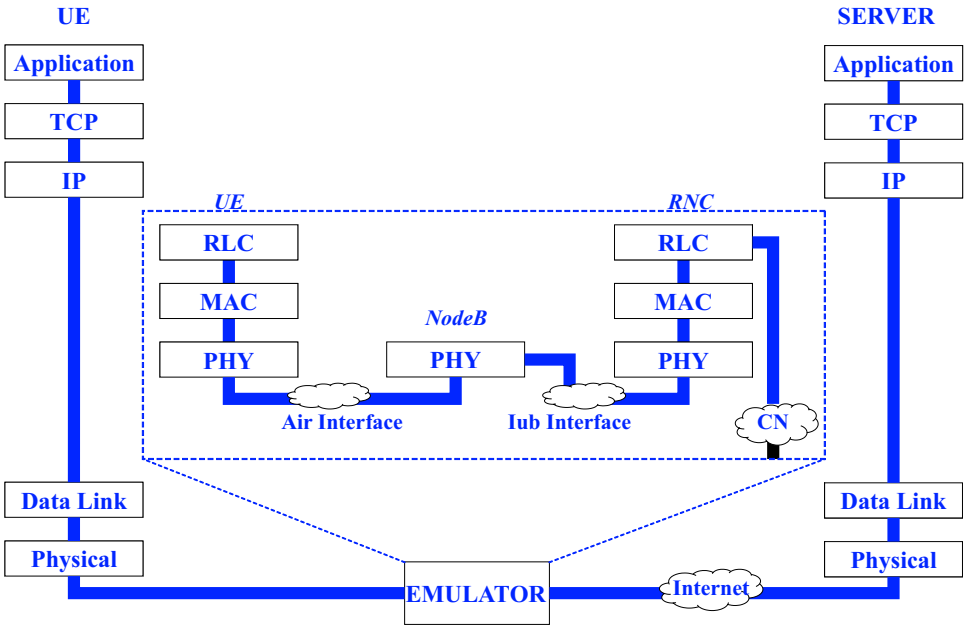



**Figure 5.9:** *Emulated UMTS protocol stack.*

ning. The emulator intercepts the packets that are coming into (both UL and DL packets), processing them according to the specified set up for the connection. Every IP packet that is arriving to the emulator is diverted into the emulated UMTS protocol stack. The RLC is fully implemented as specified in [61] by 3GPP. Specifically, IP packets are segmented into RLC PDUs, and then transmitted depending on the allocated bit rate (the MAC layer is a simplified implementation that regulates the rate of PDU transmission in order to provide a constant bit rate for the connection). The air interface is modelled to provide a uniformly distributed frame errors. Retransmissions are controlled by the polling and status reporting parameter setting specified in the configuration file. Properly received PDUs are assembled into an IP packet and forwarded to their destination.

To summarise, RESPECT is a platform that emulates a UMTS network with these main features:

- link layer segmentation, assembly, retransmission, polling and status reporting
- constant bit rate
- uniformly distributed frame errors
- processing delays of the different network entities in the E2E path
- one user
- one link

A detailed description of RESPECT, including the accuracy of emulation results generated using it, is given in Appendix B.

### 5.3.3 Parameter Settings

A combination of some of the reliability mechanisms described in section 4.2 were used. For the status reporting, *missing PDU Indication* is enabled as

it leads to the immediate detection of lost PDUs. However, if this is used, repeated status reports will be sent whenever a new PDU is received till the retransmissions of missing PDUs is received properly. To prevent these redundant status reports, different status prohibit timers are used.

The polling mechanisms are chosen in such a way as to reduce the possibility of deadlocks. Smaller set of emulation runs were undertaken to choose from these mechanisms. Poll last transmitted PDU and Poll last retransmitted PDU along with timer based polling are used in order to avoid deadlocks, as illustrated with an example in Appendix C. Different timer poll values are tested, ranging from 100 ms to 450 ms, and the results were not that different from each other. This is because the timer-based polling is mainly used to avoid deadlocks and as long as its value is not very large, the results will not differ that much. For this reason, an intermediate value of 300 ms is used for the results given in the coming section. Table 5.3 gives a summary of the most important parameter settings.

FTP file download sessions are started using automation scripts and during these sessions, it is assumed that a DCH is already set up. Thus, DCH setup and termination times are not considered. The DCH in the UL is considered to be error free, while the DL experiences a constant and uncorrelated error with an FER of 10%. The bandwidth of the DCH both in the UL and DL is fixed

**Table 5.3:** *Parameter Settings.*

| **Parameter** | **Value(s)** |
|---|---|
| maxDAT | $[1...10]$ |
| Poll last transmitted PDU | true |
| Poll last retransmitted PDU | true |
| Timer based polling (ms) | 300 |
| Missing PDU indication | *true* |
| Status Prohibit (ms) | $[50, 100, 150]$ |
| RLC Tx/Rx Window/Buffer sizes | [unlimited] |
| Bit rates (DL/UL) (Kbps) | $384/32$ |
| FER (DL/UL)(%) | 10/0 |
| TTI (ms) | 10 |
| File sizes (Kbyte) | 100 |
| TCP version | Linux 2.4 TCP with default settings [i.e. with Timestamps, SACK, FACK, DSACK enabled] |

during a download session.

For each of the considered cases, the file download is repeated 300 times. The downloads are done in a randomised fashion [73], i.e. a given case is selected randomly for the next download instead of running each case in sequence. This is done in order to distribute some systematic errors that might arise.

The results from the investigations are evaluated using the following performance metrics:

- **Download time:** The time taken to download a file. Though an FTP connection has a control and data channel [69], the time considered here is the time taken in the data channel, i.e. the time from the sending of the SYN packet in the data channel till the arrival of the last data packet in the data channel.

- **Throughput:** Throughput is the average amount of data that is transmitted in a given time, including retransmissions and status reports. Two different types of throughput measures are used here, *Mean peak throughput* and *Moving average throughput*. The mean peak throughput is calculated by averaging the throughput during each TTI (i.e bits sent per TTI divided by one TTI duration). On the other hand, the moving average throughput is calculated by dividing the total number of bits sent during that session by the total time spent.

- **RTT:** The average time elapsed from the arrival of a TCP packet at the sender till its removal from the transmission buffer due to the reception of an ACK signifying its complete reception.

- **TCP retransmission ratio:** The percentage of TCP packets that are retransmitted.

- **Status Overhead:** The ratio of status PDUs that are sent in the UL to the data PDUs that are sent in the DL. This indicates how much penalty is paid in UL capacity for the sake of reliability in the DL.

### 5.3.4 Results and Discussion

Figure 5.10 shows the dependency of the download time on maxDAT and status prohibit. As can be seen from the figure, for a given status prohibit value, the download time increases as the maxDAT value decreases. For the status prohibit values of 50 ms, 100 ms and 150 ms, increasing the maxDAT value from 1 to 2 leads to a decrease in the download time of 56%, 56% and 79% respectively[3]. This is an expected result as the main idea behind link layer retransmissions is to decrease the probability of TCP timeouts by retransmitting a subset of the packet, i.e. RLC PDUs.

This is illustrated in Figure 5.11, which shows the TCP retransmission ratio. It can be seen that, the download times follow the same pattern as the TCP retransmission rate for maxDAT values less than 4. The TCP retransmission ratio is around 33%, even though the FER is only 10%, for all the three cases when maxDAT is 1. This is because for a bit rate of 384 Kbps, a 1500 byte IP packet needs:

$$\frac{1500 * 8 \frac{bits}{packet}}{384000 \frac{bits}{sec} * 0.01 \frac{sec}{block}} = 3.12$$

radio blocks to be transmitted, and if any one of this gets lost the packet is also lost as there is no retransmission, hence raising the TCP retransmission ratio to approximately three times that of the FER. For maxDAT values greater than or equal to 4, TCP timeouts are prevented almost completely.

The effect of maxDAT becomes marginal when its value is greater than 4, because the probability of having more than 3 retransmissions is negligible (as shown in Figure 5.1 in Section 5.2.1). For lower values of maxDAT, status prohibit has an important effect on the download time: smaller status prohibit values lead to longer download times, and vice versa. This is because, lower

[3]Setting maxDAT to 1 is equivalent to disabling retransmission, and hence nullifying the RLC reliability mechanism

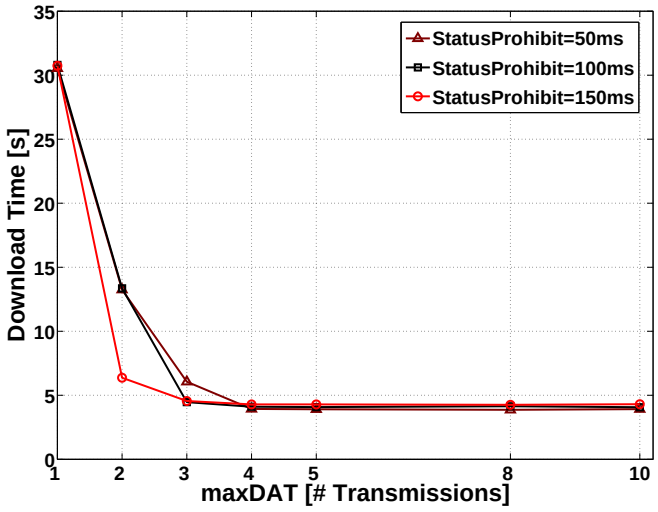



**Figure 5.10:** *Mean file download times for different maxDAT and status prohibit values, for a 100 Kbyte file.*

status prohibit timers mean higher status reporting frequency, increasing the probability that a PDU is retransmitted. An increase in the retransmission ratio will make the retransmission count reach the maxDAT faster, and the SDU will be discarded. The SDU discard will lead to TCP timeouts, and hence

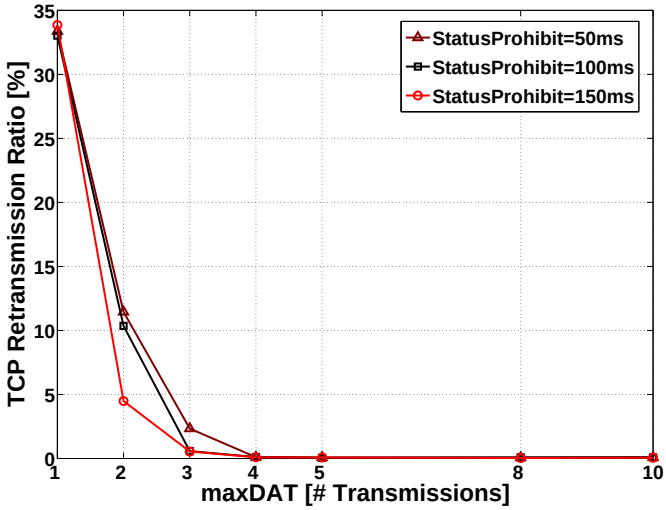



**Figure 5.11:** *Mean TCP retransmission ratio for different maxDAT and status prohibit values, for a 100 Kbyte file.*

decrease the link efficiency. However, for larger maxDAT values ($\geq 4$), even though smaller status prohibit values lead to spurious RLC retransmissions as too many status reports arrive due to missing PDU detection, maxDAT is high enough to prevent the untimely discard of an SDU. Actually, there is even a slight improvement in the download time (in the range of 0.4 seconds) when the status prohibit is reduced from 150 to 50 ms for larger values of maxDAT.

Though lower status prohibit values lead to lower download times when the maxDAT is high enough, the down side is that more status reports will be sent in the UL direction, i.e there is an increase in status overhead for low status prohibit values. This is illustrated in Figure 5.12. The values are shown starting from a maxDAT value of 2 because when maxDAT is 1, there is no retransmissions (except for the sake of polling), and the status reporting is not useful at all. As shown in the figure, the status overhead decreases with maxDAT because the chances that a status report is wasted (i.e. a status report asking a retransmission is sent by the UE but no retransmission results out of it) becomes smaller with increasing maxDAT.

A higher status overhead is a disadvantage as it means that more UL capacity is required. For example for maxDAT value of 2 and a status prohibit value of 50 ms, the status overhead is around 5% (the maximum value for all the cases shown), which means approximately 19.2 Kbps ($384 \times 0.05$) UL ca-

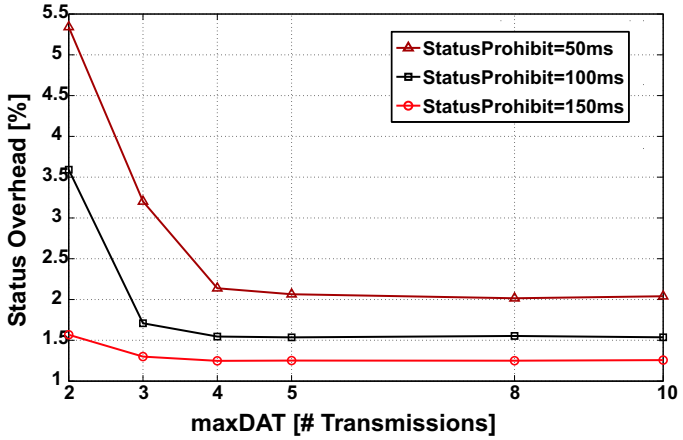



**Figure 5.12:** *Mean Status Overhead for different maxDAT and status prohibit values, for a 100 Kbyte file.*

pacity is required. Fortunately, only 10.2 Kbps ($\frac{384\times40}{1500}$) is required for sending ACKs at the TCP level if the maximum throughput is reached. Thus, even in the worst scenario of requiring 19.2 Kbps for status reporting, 32 Kbps is still enough to provide the maximum 10.2 Kbps required for ACKs.

Figure 5.13 shows the mean RTT for the different cases. As can be seen from the figure, the RTT is very low for small values of maxDAT. This is because when maxDAT is very low, there are a lot of SDUs that are discarded and the only SDUs that will contribute to the mean RTT calculation are the ones that are completed with fewer PDU being retransmitted. For example, for the maxDAT value of 1, the only SDUs that are taken into account for the RTT calculation are the ones that are received without any of their PDUs being retransmitted. When maxDAT increase, so does the RTT as more and more SDUs are being received properly, mostly with some of their PDUs having being retransmitted. After maxDAT reaches 4, the RTT value remains stable. With low status prohibit values, retransmission requests arrive to the sender in a quick succession, increasing the rate of retransmission of PDUs, and hence decreasing the total time required to transmit a given SDU.

Figures 5.14 and 5.15 show the mean peak and average throughput, respectively. From the figures it can be seen that the throughput follows a trend similar to the download time shown in Figure 5.10. That is, lower status prohibit

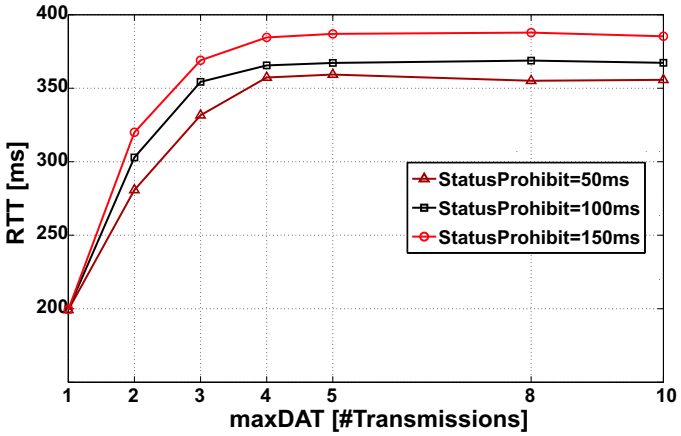



**Figure 5.13:** *Mean RTT for different maxDAT and status prohibit values, for a 100 Kbyte file.*

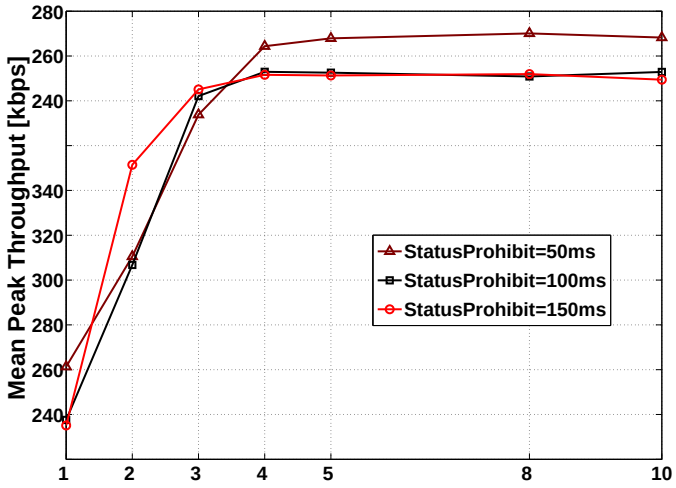



**Figure 5.14:** *Mean peak throughput for different maxDAT and status prohibit values, for a 100 Kbyte file.*

values work better at larger maxDAT and vice versa, and also that performance improves with increasing maxDAT. It should be noted that the direct parallel that is found here between download time and throughput is not a necessity, because theoretically it is possible to have a high throughput that also leads to

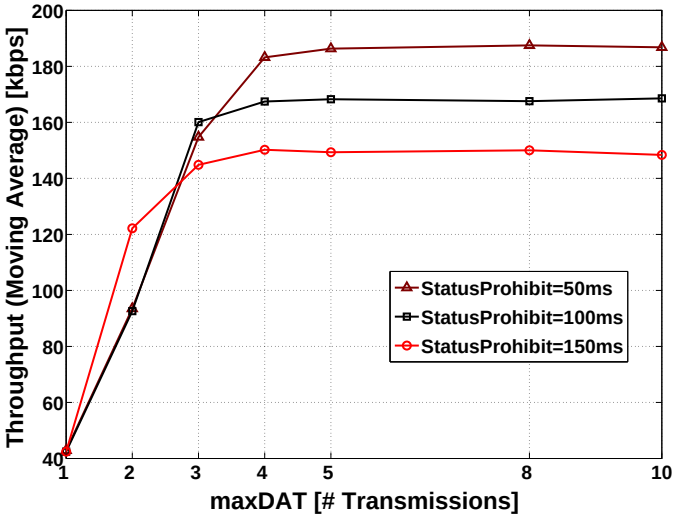



**Figure 5.15:** *Mean moving average throughput for different maxDAT and status prohibit values, for a 100 Kbyte file.*

longer download times, if most of the throughput was wasted on unnecessary retransmissions. This again shows the effectiveness of RLC's retransmission mechanism. The effect of TCP's slow start can also be seen from Figure 5.15, where the maximum average throughput achieved is less than 200 Kbps, even though the available bandwidth is 384 Kbps.

Figure 5.16 shows the retransmission delay. When maxDAT is 1, the retransmission delay for all cases is around 300 ms. This is because when the maxDAT value is 1, there is no retransmission in effect, and the only reason a retransmission can be allowed is in the case of a poll is needed to be sent and a status is not received when the poll timer has expired [74; 61]. If there is no new PDU to be sent at that time, one of the old PDUs still waiting for an ACK will be retransmitted just for the sake of polling, and hence leading to a retransmission delay equal to the poll timer value, i.e. 300 ms. When maxDAT becomes greater than 1, the retransmission delay stabilises and as expected, the larger the status prohibit timer, the larger the delay. The corresponding retransmission delay values for the status prohibit timers of 50,100, and 150 ms are 180, 200, and 220 ms, respectively.

Finally, the emulation results are compared with the theoretical model de-

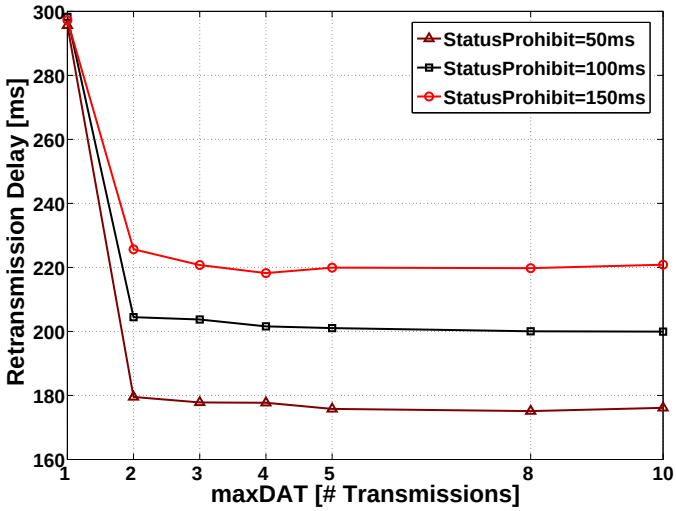



**Figure 5.16:** *Mean retransmission delays for different maxDAT and status prohibit values, for a 100 Kbyte file.*

veloped at the beginning of this chapter in order to validate the emulations. Figure 5.17 shows the comparison between the actual download times from the emulation results and the theoretical values evaluated using (5.14) in Section 5.2.2. Since the theoretical analysis did not consider the TCP connection setup and termination times for the data channel, the values are adjusted here to include the time taken for this procedure by adding $2 \times RTT$, one for the connection setup and one for the termination. An RTT is added instead of $1.5 \times RTT$ (as discussed in Chapter 3) for the connection setup and termination because the data channel setup procedure is initiated by the server and hence, the data will start flowing immediately after the server receives the ACK for the SYN. The maximum relative error between the theoretical and actual values is within 9%.

Figure 5.18 shows the comparison between the actual RTT from the emulation results and the theoretical values evaluated using (5.8) in Section 5.2.2. As can be seen there is more difference between the theoretical and emulation RTT values (up to 17%) as compared with the download time comparison. This is because in the theoretical estimation, the impact of queueing was not considered.

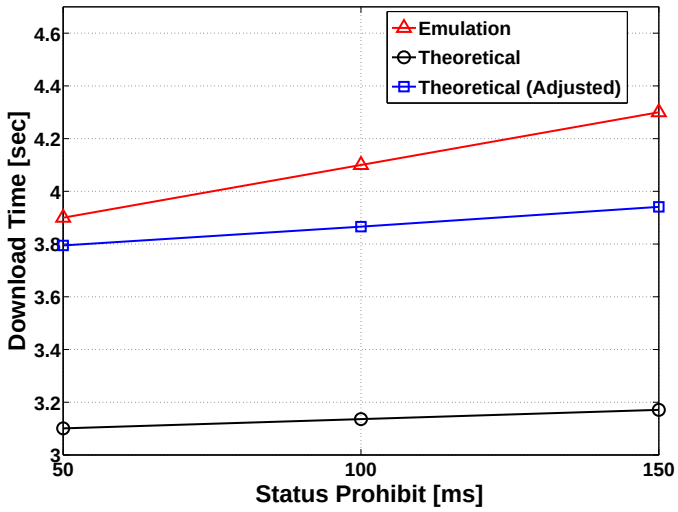



**Figure 5.17:** *Comparison of actual and theoretical download times.*

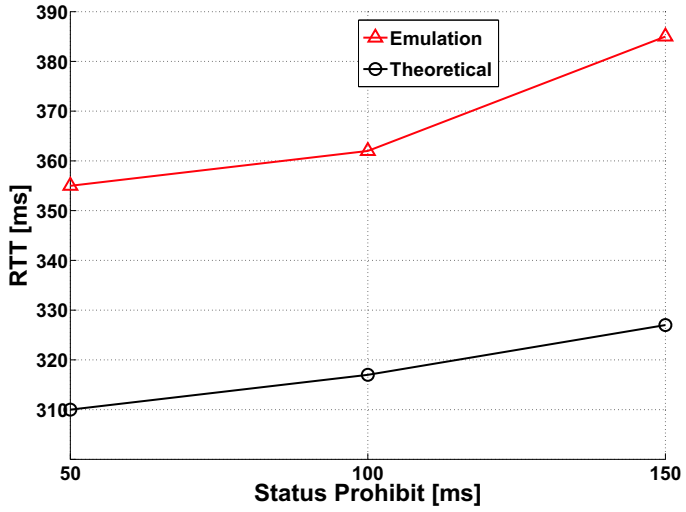



**Figure 5.18:** *Comparison of actual and theoretical RTT.*

All the results shown in this chapter were generated for a 100 Kbyte file download. The performance impact on larger files will be similar to the 100 Kbyte case because, as discussed in Section 5.2.2, TCP reaches its steady state for the considered settings very quickly. If very small file sizes are considered the performance trend will still remain the same but the differences will be more pronounced. For example, if a 1 Kbyte file is considered, only one packet is needed to transfer the file and using a maxDAT configuration that might result in a TCP timeout could increase the download time significantly.

## 5.4 Summary

In this chapter, the performance of FTP file download over a UMTS dedicated channel, under the assumptions of constant bit rate and uncorrelated errors has been investigated. The investigations has been carried out using simplified analytical modelling and detailed real time emulations using a standard TCP implementation and a UMTS network emulator.

It is found that the main factor determining the performance is the maxDAT

value, higher values giving the best goodput. The only downside of increased maxDAT is increased RTT, but for the FTP service considered here that is not an issue. However, for services such as web browsing where the RTT is also an important performance metric, intermediate values of maxDAT are recommended.

Status prohibit plays a significant role when the maxDAT is not large enough, values closer to the RTT of the UMTS network giving better performance. When the maxDAT is sufficiently large ($\geq 4$), status prohibit has minor influence on performance: lower status prohibit values leading to performance improvement. However, the improvement is inefficient because the big throughput increase contributes only marginally to the goodput.

The status overhead is shown to be very low, with a maximum value of around 5%. In other words, the UL capacity required to accommodate status reporting is small, showing that the retransmission mechanism employed by RLC is very efficient.

# Chapter 6

# Packet Service Performance in UMTS: Impact of Spurious Retransmissions

In the previous chapter, the impact of different RLC parameters on the performance of an FTP download session was discussed. The default TCP settings of the Linux operating system were used, as the focus was solely on the RLC protocol. In this and the next chapter, the focus shifts to TCP.

As discussed in the previous chapter, TCP's performance in UMTS can be improved by enabling persistent link layer retransmissions. The link layer retransmissions were shown to transform the unreliable wireless channel into a reliable one with variable delay. Unfortunately, higher channel errors is not the only problem, and wireless networks can suffer more than wired networks from spurious retransmissions which are caused by several factors such as a sudden increase of delay in the communication path. In this chapter we investigate the impact of these spurious retransmissions on TCP performance.

The chapter is organised as follows. We start by discussing the causes of spurious retransmissions in UMTS and some recommended solutions in Section 6.1. Section 6.2 gives a brief survey of related work. The investigated

cases and evaluation results are discussed in Sections 6.3 and 6.4, respectively. Finally, Section 6.5 summarises the main findings from the investigations.

## 6.1 Spurious Retransmissions

### 6.1.1 Causes of Spurious Retransmission

A *spurious retransmission* is an unnecessary retransmission that is performed by TCP due to a mistaken inference of packet loss. The main causes of spurious retransmissions are *packet reordering* and *delay spikes*.

**Packet Reordering** In wired networks, as different packets can traverse the network via different paths, packets may be out of sequence when they reach the receiver. As mentioned in Chapter 3, TCP responds to out of order reception by sending duplicate ACKs, which might trigger a *spurious fast retransmit*. Measurements performed in [75], [76] and [77] show that packet reordering is not uncommon on the Internet, and the probability of a reordering happening in a TCP session can be as high as 90%, depending on the network load.

As mentioned in the previous chapter, the RLC receiver in UMTS assembles SDUs and sends them to upper layers in or out of sequence depending on the setting of the *in-sequence delivery* option. Thus, on top of the reordering that might happen in the wired part of the E2E path, RLC out of sequence delivery will result in reordering at the TCP level as demonstrated in Figure 6.1.

**Delay Spike** A *delay spike* is a sudden increase in the latency of a link and it is more prevalent in wireless networks than in wired ones. The main causes of delay spikes are bandwidth downgrade, link outage that is masked through persistent link layer retransmission, handover, and blockage by high priority traffic [24; 78]. The packets that were delayed will arrive after the delay spike,

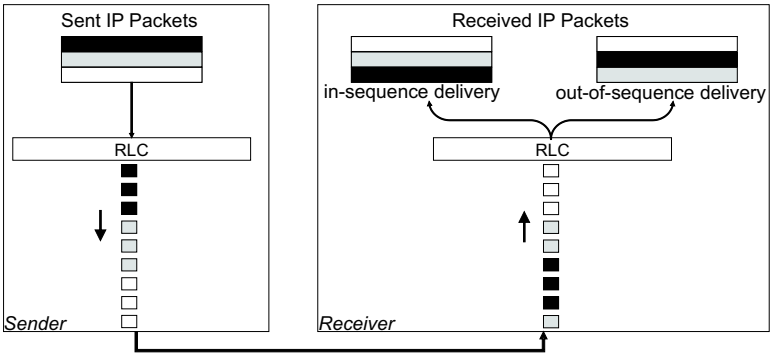



**Figure 6.1:** *Out of sequence delivery of RLC leading to packet reordering at TCP level.*

but TCP might have already timed out (*spurious timeout*) and retransmitted the packets.

### 6.1.2 Impact on Performance

In [75], and [76], investigations were carried out on the impact of packet reordering on TCP performance, the results showing that reordering leads to performance degradation by making TCP resort to too many fast retransmit/recovery invocations. Duplicate reception of retransmitted packets can also further trigger spurious fast retransmissions due to the reception of multiple duplicate ACKs. Thus, spurious fast retransmissions lead to bandwidth under-utilisation due to unnecessary retransmissions and unnecessary reduction of the congestion window.

Figure 6.2 illustrates the impact of spurious timeouts from a trace of a TCP connection. Around 1.5 seconds after the connection has started, a delay spike of 4 seconds was artificially introduced. The packet that caused the timeout was retransmitted twice, first at around 2.8 seconds, and a second time at 5.4 seconds (due to the exponential back-off procedure described in Chapter 3). The delay spike is finished at around 5.5 seconds, and the receiver starts sending an ACK for the reception of the original transmissions. In ordinary TCP[1],

[1]In this report, the term “ordinary TCP” refers to a non SACK new-Reno implementation of TCP.

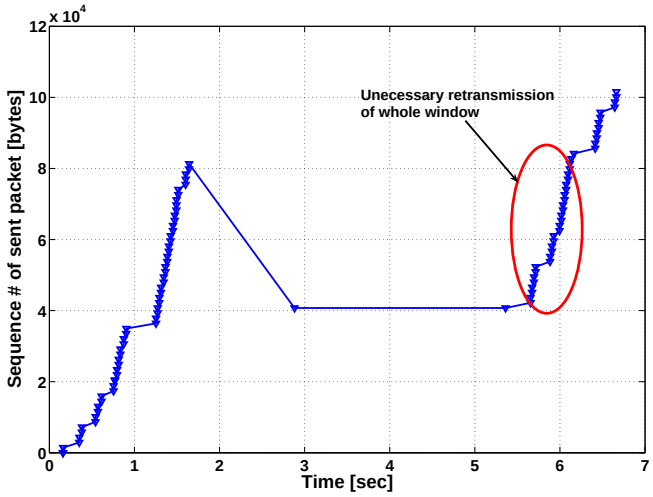



**Figure 6.2:** *Example trace showing unnecessary retransmission of the whole window of outstanding data due to TCP's RTO recovery.*

there is no way to distinguish between an ACK that is due to retransmissions from an ACK that is acknowledging original transmission [79]. This is known as the *retransmission ambiguity* problem of TCP. Therefore, TCP can not tell whether the retransmission it is performing are valid or redundant and it assumes the incoming ACKs are in response to the retransmissions and will keep on retransmitting the rest of the window.

### 6.1.3 Possible Solutions

The timing out of the retransmission timer due to delay spikes is hard to prevent, as the sender is not aware of the delay spikes in the communication path. In [80], a scheme is introduced where delays are artificially injected on the communication path in a certain fashion to increase the RTT variance estimate and hence avoid timeouts due to delay spikes. However, this scheme is not effective if the delay spikes are long.

Fortunately, something can be done after the delay spike is over and the ACKs for the original transmissions start to arrive. A solution for the spurious retransmission problem has first to detect if the retransmissions are spurious,

and if so revert unnecessary congestion control phases such as slow start that the sender have mistakenly entered after the spurious retransmission.

Some of the well known proposals for detecting and responding to spurious TCP retransmissions are:

**Eifel Algorithm:** The timestamps option [50] can be used to detect spurious retransmissions. When the timestamp option is used, the TCP sender inserts the time of the packet generation in the TCP header and when ACKs are generated for these packets, the timestamps are echoed back. Thus, TCP will be able to know whether incoming ACKs are due to first transmissions or retransmissions. The *Eifel* algorithm proposed in [26; 81] uses the timestamp for the detection of spurious retransmissions[2]. When a congestion is detected (either through triple duplicate ACKs or a timeout), Eifel saves the congestion parameters along with the timestamp of the retransmission sent out. If the ACK received later on for the concerned packet echoes back an older timestamp than the saved one, the retransmission was a spurious one and Eifel undoes the congestion procedure and reverts to the state just before the spurious retransmission.

**DSACK:** As described in Section 3.3, TCP DSACK provides a possibility to report duplicate reception of data and this can be used to detect spurious retransmissions. In [82; 83; 84] a detailed description of how DSACK can be used to detect spurious retransmissions is given. When congestion is detected, the current congestion parameters are saved and the normal congestion procedure is undertaken. If a DSACK arrives signifying duplicate reception, the congestion control procedure is undone by restoring the saved parameters. Note that Eifel and DSACK can coexist in the same TCP stack and be enabled at the same time (this is the case in Linux TCP) as their operations are independent from each other.

[2] Instead of using timestamps, a reserved bit in the TCP header can also be used in Eifel as suggested in [26].

**Forward RTO Recovery (FRTO):** Spurious retransmissions can also be detected without using DSACK or timestamps. FRTO is one such TCP enhancement [85; 86]. When a retransmission timeout occurs, FRTO retransmits the packet without going into slow start (the ssthresh is, however, set to half of the outstanding packets). If the first ACK after the timeout advances the window, FRTO transmits two new packets instead of retransmitting the other packets in the same window and enters congestion avoidance (i.e cwnd is set to ssthresh). If the second ACK also advances the window, the retransmission was most likely spurious because the second ACK is triggered by an originally transmitted packet. Otherwise, if either of the first or the second ACK is a duplicate ACK, this is an indication of a correct retransmission and FRTO will go into slow start and continue as in ordinary TCP.

## 6.2 Related Work

Previous studies with regards to spurious retransmissions in wireless networks mainly dealt with delay spikes only. In [87], the Eifel algorithm is evaluated in a simulated GPRS network with frequent cell re-selections occurring that cause delay spikes. It is shown that Eifel can reduce the download time by up to 12%. It is also shown that Eifel can lead to performance degradation when real losses occur during the delay spike.

In [88], it is shown that Eifel performs better than TCP Reno in low bandwidth links with no packet loss, while its performance worsens for high bandwidths with packet loss. Similar results are reported in the studies carried out in [89].

In [90], an enhanced version of the Eifel algorithm[3] is suggested and its performance is evaluated in a high bandwidth (2 Mbps) wired link that suffers from frequent delay spikes during a 5 Mbyte file download session. It is shown that Eifel can increase throughput by up to 240% in a non congested link, but

[3]The enhancements include restoring the congestion window even during multiple timeouts, and complex RTO calculation that takes the frequency of RTT samples into consideration.

can lead to performance degradation if a congestion is also occurring along with the delay spikes.

Previous packet reordering studies mostly are concerned with wired networks. The author is aware of only one published work, described in [91], that dealt with reordering due to link layer ARQ and its impact on TCP. Therein, the simulations with TCP new Reno showed that using out of sequence delivery can lead upto 50% reduction in the throughput.

The studies carried out in this chapter differ from the ones mentioned above in three main aspects, apart from issues of the realistic context of the settings mentioned in Section 5.1. First, the focus here is mainly on packet reordering though delay spikes are also considered. Secondly, a combination of several TCP versions are studied. Finally, short-lived flows are investigated instead of the bulk file downloads[4].

## 6.3 Investigated Scenarios

As in Chapter 5, RESPECT is the tool used here to perform the investigations (Appendix B). The investigations are divided into two parts: effect of out of sequence delivery and delay spikes[5]. For both cases, the performance of different TCP extensions are investigated.

The RLC parameters that deal with retransmissions are fixed based on the results from the previous chapter, and the only RLC parameter that is investigated here is the in-sequence delivery. Though there are several TCP extensions and optimisation proposals, only a few of them are well accepted by the research community to be standardised by organisations such as the Internet Engineering Task Force (IETF). Apart from that, many of the proposals that are targeted to wireless networks assume a non reliable link, which does not

[4] Studies carried out in [92] indicate that most Internet flows are short lived (ranging from few seconds to few minutes), and we assume the traffic in the mobile Internet will be even shorter due to cost and bandwidth limitations.

[5] For the delay spike investigations, RESPECT was slightly modified to be able to introduce arbitrary delays at any time during a TCP connection.

apply to UMTS as the RLC protocol provides reliability. Most of the standardised extensions/options are not even used in many hosts on the Internet [93; 94]. In [94] investigations were carried out where some of the most widely accessed web servers were probed to find out their TCP capabilities. The results showed that except for SACK, which more than 70% of the servers were found to support, most of the other TCP extensions like limited retransmit have very low deployment. In [95], it is found that 76% of the major Internet web servers use the timestamp option.

With this in mind, we limit our focus here to those options/extensions that have impact on spurious retransmissions and which are currently being standardised. Also, from these we consider only the ones that are currently available in Linux, as Linux is the most widely used operating system for hosting web servers [96]. In Linux, the concept of spurious retransmission detection is built in the TCP implementation [54]. Spurious retransmissions are detected either through timestamps or DSACK. Once a spurious retransmission is detected, congestion parameters are reverted to the state just before the retransmission.

FTP file download sessions were started using automated scripts under the assumption that a DCH is already setup. For each of the considered cases, the file download is repeated 300 times, and as in the previous chapter, the downloads are done in a randomised fashion. Table 6.1 summarises the different settings used for the investigations in this chapter[6].

## 6.4 Emulation Results

### 6.4.1 Impact of Delivery Order

Figure 6.3 shows the impact of the RLC delivery order on the download time under different TCP settings. As can be seen from the figure, when in-sequence delivery is used, there is no significant difference ($< 1\%$) in the download

[6] Note that FRTO is not considered, as the Linux implementation of FRTO was found out to have bugs [97].

**Table 6.1:** *Parameter Settings for reordering and delay spike investigations.*

| Parameter | Value(s) |
|---|---|
| *Common Settings* | |
| maxDAT | 10 |
| Poll last transmitted PDU | true |
| Poll last retransmitted PDU | true |
| Timer based polling (ms) | 300 |
| Missing PDU indication | *true* |
| Status Prohibit (ms) | 100 |
| Bit rates (DL/UL) (Kbps) | 384/32 |
| TTI (ms) | 10 |
| File sizes (Kbyte) | 100 |
| Eifel (Timestamps) | on/off |
| TCP SACK | on/off |
| TCP FACK | on/off |
| TCP DSACK | on/off |
| TCP FRTO | off |
| File Size [Kbyte] | 100 |
| FER (DL/UL)(%) | 10/0 |
| *Reordering Investigations* | |
| RLC delivery order | in sequence, out of sequence |
| *Delay Spike Investigations* | |
| RLC delivery order | in sequence |
| Extra delay duration (sec) | [0.5, 1, 2, 3, 5] |
| Extra delay location (sec) | [0, 0.5, 1, 2] |

times between the different TCP settings. This is because no packets are delivered out of order, i.e. no fast retransmissions due to triple duplicate ACKs. The small differences between the Eifel and non-Eifel cases for the in-sequence delivery is due to the overhead of including the timestamps in the TCP header, which requires 10 additional bytes in the TCP header [50]. Henceforth in this section, unless otherwise specified, the discussions refer to the out of sequence cases.

For out-of-sequence delivery, when no SACK or its enhancements are used, using Eifel decreases the download time by 33%, which translates to a 68 Kbps increase in the goodput. The use of SACK/FACK increases the download time, for both the Eifel and non Eifel cases, but by no more than 5% (< 10 Kbps). When Eifel is not used, the use of DSACK decreases the download time by 14%, corresponding to a 20 Kbps improvement in the goodput.

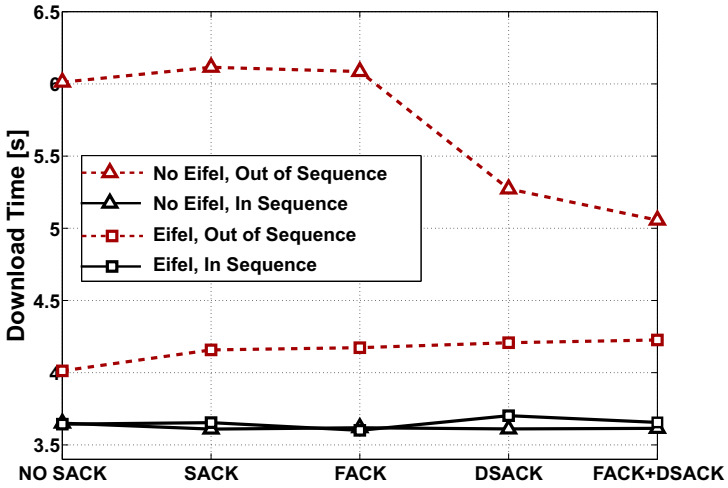



**Figure 6.3:** *Impact of RLC delivery order: Mean download times.*

In order to understand these differences, the TCP retransmission ratio is plotted in Figure 6.4. When Eifel is not used, using SACK reduces the retransmission ratio from 13% to 9% while it increases the download time from 6.0 seconds to 6.12 seconds. The reason why SACK leads to less retransmissions is because the sender will not retransmit already received packets (as

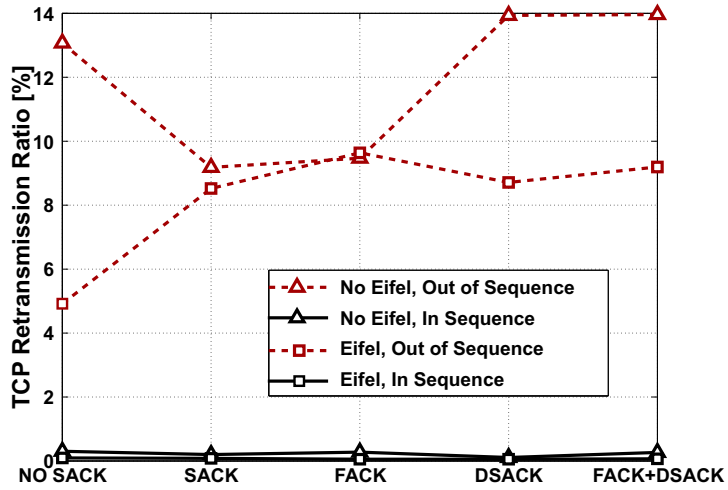



**Figure 6.4:** *Impact of RLC delivery order: Mean retransmission ratios.*

they are indicated in the SACK blocks), while it might do so in the non SACK case. The fact that download time is increasing while the retransmission ratio is decreasing looks contradictory at first look, but it becomes clear when we consider not only the number of retransmissions but also the time when these retransmissions occur.

In Figure 6.5, the Cumulative Distribution Function (CDF) of the time instant at which the first retransmissions were sent for the SACK and no SACK cases are shown. As can be seen from the figure, using SACK, regardless of the Eifel setting, leads to the first retransmission to happen earlier.

The reason for this earlier retransmission behaviour of SACK is because Linux's implementation of SACK TCP is more aggressive than non SACK TCP when it comes to entering fast retransmit and recovery phase. In non SACK TCP fast retransmission is performed only when a certain number of (usually three) duplicate ACKs are received, while in SACK TCP fast retransmission can be performed if a SACK block arrives that acknowledges a packet that is three or more packets away from the last packet that was cumulatively

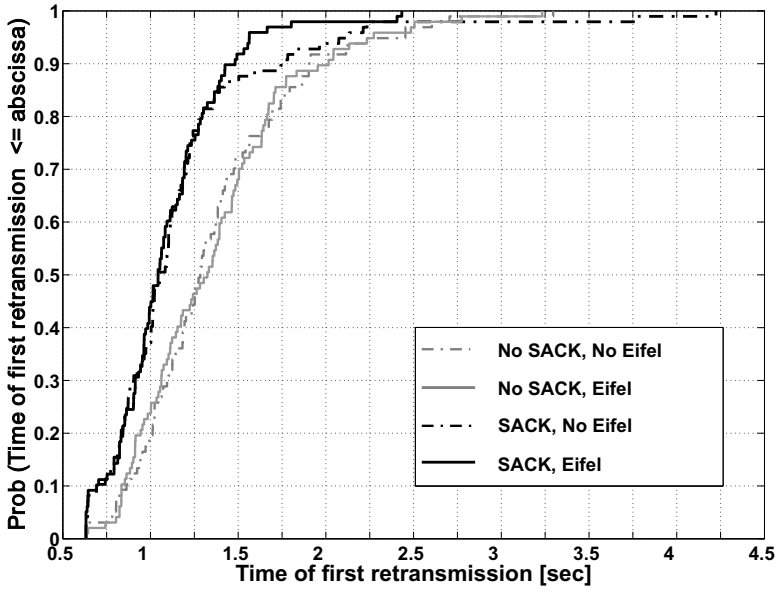



**Figure 6.5:** *Impact of RLC delivery order: CDF of the time of the first TCP retransmission.*

ACKed[7].

This is illustrated in Figure 6.6. The figure shows the trace of the sent packets (the small rectangles), and ACKs received (solid curve representing cumulative ACKs and vertical bars representing the start and end of the SACK blocks) at the sender. Retransmissions are represented by a cross. At around 0.8 seconds, there is a retransmission because three duplicate ACKs are received. However, at around 1.1 seconds, there is a retransmission even though only one duplicate ACK is received because the SACK block that was received (acknowledging bytes 29524-30984) along with the duplicate ACK was 4 packets (4*1460 bytes) away from the duplicate ACK (acknowledging byte number 23684). Thus, TCP SACK in Linux acts like TCP FACK (described in Section 3.3) when it comes to triggering fast retransmissions[8] [98].

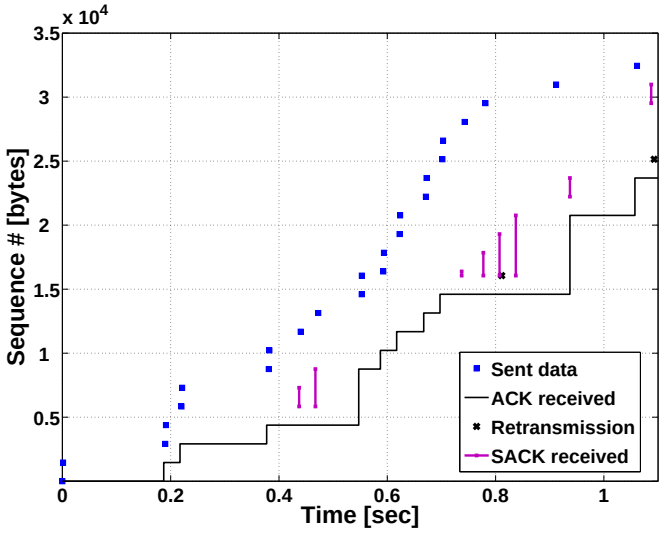



**Figure 6.6:** *Time sequence graph illustrating Linux TCP SACK's fast retransmission behaviour.*

[7] In Linux, the initial value of the number of duplicate ACKs required to trigger fast retransmission is three. However, when reordering is detected, this threshold is updated to the number of packets between the highest acknowledged packet so far and the currently acknowledged packet that was found to be duplicated [54].

[8] However, the behaviour is different during recovery because TCP FACK assumes the holes between the SACK blocks as lost, while in SACK they are assumed to be lost only if three discontiguous blocks are SACKed above the concerned packet or if a packet more than three packets away from the concerned packet is SACKed [58].

The earlier fast retransmissions would have been beneficial if there was actual data loss, but there is none as persistent RLC link layer retransmission is employed. The end result of the earlier fast retransmits is to drive TCP into congestion avoidance earlier than non-SACK TCP. Hence, the reason for the increase in the download time despite the decrease in the retransmission ratio when using SACK without Eifel.

Returning to Figure 6.3 and 6.4, it can be seen that when Eifel is used, SACK leads to an increase in the download time and also the retransmission ratio. Actually, the retransmission ratio is almost the same for the SACK/FACK case regardless of the Eifel setting. This is because the retransmission behaviour of TCP SACK described earlier is prevalent. However, in the no SACK case, TCP is able to detect unnecessary retransmissions with the help of timestamps and stop from further unnecessary retransmissions of the whole window. Thus, there is a reduction in the retransmission ratio when Eifel is used without SACK as compared with the case when it is used with SACK. FACK performs basically the same as in SACK because the early fast retransmit behaviour is the same and also the FACK algorithm is disabled in Linux TCP if reordering is detected [54].

Figures 6.3 also shows that for the case where Eifel is not used, DSACK leads to a 13% improvement in the download time. The reason for the decrease in the download time is obvious, because with DSACK, spurious retransmissions can be detected and hence, unnecessary fast recovery phases are reverted. However, the improvement is less than the one we can get from using Eifel because a DSACK block is sent in response to the reception of the unnecessary retransmission while an ACK arriving with a timestamp earlier than the retransmission's timestamps immediately after the retransmission is sent out will give the same information. Thus, Eifel is able to recover from spurious retransmissions faster than DSACK.

Using FACK on top of DSACK, when Eifel is not used, leads to around 5% further decrease in the download time. This is because FACK is aggressive when it comes to retransmissions during fast recovery, as it considers all the holes between the SACKed blocks as lost. This increases the possibility of

duplicate packet reception in the case of spurious retransmissions, and as such decreases the wait time for a DSACK block, which shortens the unnecessary fast recovery period.

Figure 6.7 shows the RTT for the different cases. As expected, the out of sequence delivery cases lead to lower RTT as a packet is released immediately after it is assembled properly by the RLC, while in the in sequence delivery cases there is a re-sequencing delay. Also, note that Eifel leads to increased RTT for both cases, leading to a 42% increase, on the average, for the out of sequence cases and only a 3% increase for the in sequence cases. The reason for this increase is due to the retransmission ambiguity mentioned in Section 6.1.2. That is, when timestamps are not used, it is not possible to know whether incoming ACKs are for original packets or for retransmissions and thus they are not considered in the RTT statistics. And for the cases considered here, as there were no packet drops, retransmitted packets are the ones that were delayed due to out of sequence delivery, and as such are the ones with the highest RTT value.

As all the results presented so far in this section show, out of sequence delivery leads to performance degradation in terms of download times, but it

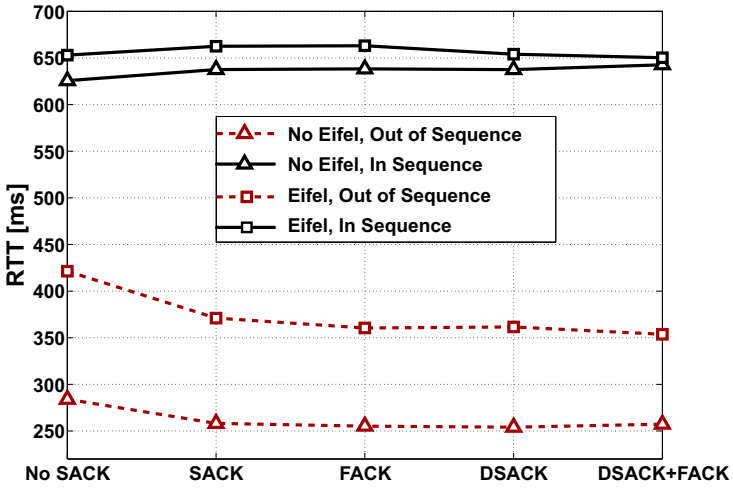



**Figure 6.7:** *Impact of RLC delivery order: Mean RTT.*

leads to a lower RTT. Another advantage of out of sequence delivery is the lower memory requirement for buffering. Figure 6.8 shows the CDF of the maximum buffer occupancy experienced at the UE during a download session. As can be seen from the figure, the buffering requirement of in sequence delivery is roughly twice that of out of sequence delivery. The absolute buffering requirement is not that much considering that even GPRS phones such as the Nokia 6600$^{TM}$ come with megabytes of built-in memory and the price of flash memory is becoming very cheap [99].

### 6.4.2 Impact of Delay Spikes

Figure 6.9(a) shows the TCP retransmission ratio for delay spikes of different durations averaged over the different delay locations listed in Table 6.1. As expected, as the delay duration increases, so does the number of retransmissions. However, the percentage increase decreases for larger delay values mainly due to the exponential back-off mechanism of TCP described in Section 3.2.3. For example, if a timeout occurs when the RTO is 3 seconds, the RTO will be doubled and the next retransmission will occur after 6 seconds. For example,

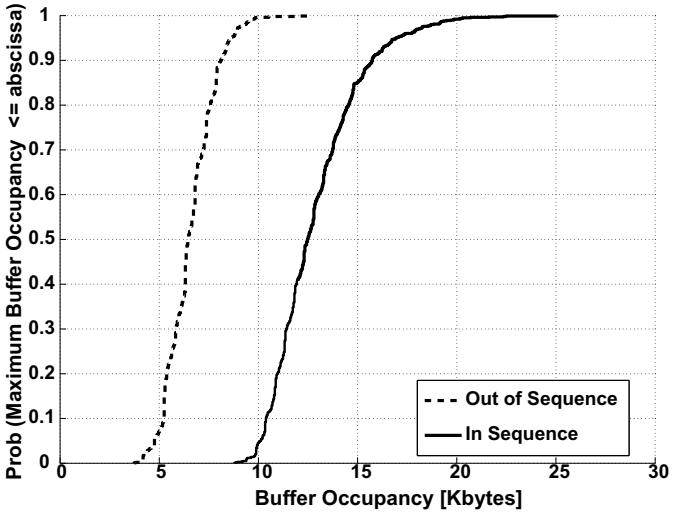
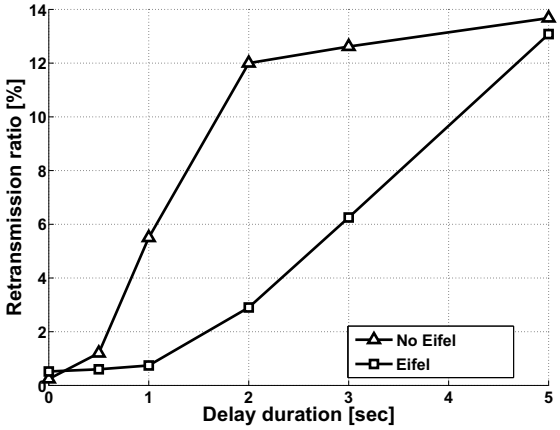
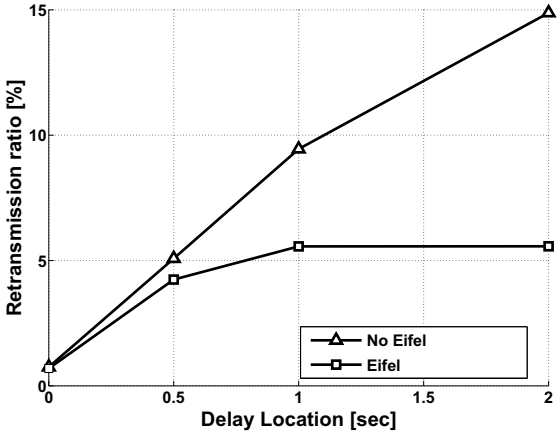



**Figure 6.8:** *Impact of RLC delivery order: CDF of maximum buffer occupancies at the UE.*




*(a)Impact of delay duration*




*(b)Impact of delay location*

**Figure 6.9:** *Impact of delay duration and location: Mean retransmission ratios.*

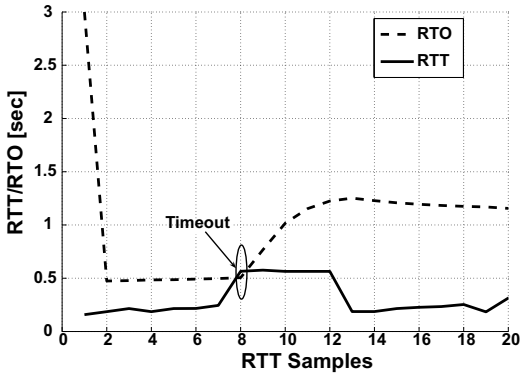
a total delay value of 5 seconds will not cause any more retransmissions than that of a 4 second delay for this case.

From Figure 6.9(a), it can also be seen that using Eifel reduces the retransmissions rates. However, the relative improvement decreases as the delay duration increases. This is because when multiple timeouts occur, the Eifel algorithm stops the congestion window reversal behaviour [26].

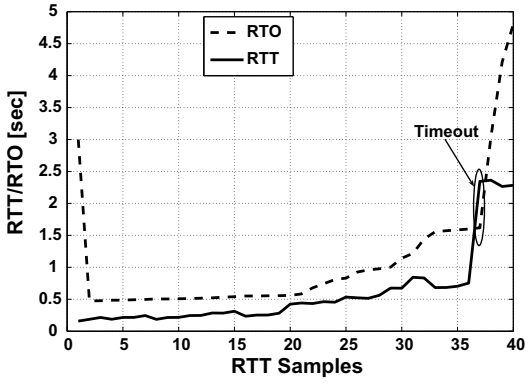
Figure 6.9(b) shows the TCP retransmission ratio for delay spikes applied at different locations averaged over the different delay durations. When the

delay is applied at 0 seconds, the SYN packet is the one that is affected, and the RTO estimator will not get any RTT samples to calculate the next estimate and the RTO stays at 3 seconds. So delay values less than 3 seconds do not trigger any retransmissions, hence the average number of retransmissions when the different delays are applied at the beginning of the connection is very low.

From Figure 6.9(b) it can also be seen that when the delay location increases, so does the retransmission rate, but the percentage increase decreases. This is because when the delays are applied earlier, even small delay durations can cause a timeout. This is better understood by looking at sample traces, shown in Figure 6.10. The RTO curves are generated by using the RTO estimation algorithm employed in Linux [100]. Since the initial RTT experienced by the first few packets is very low compared to the 3 second initial RTO, the




*(a)Delay location = 0.5 sec*




*(b)Delay location = 2 sec*

**Figure 6.10:** *Example traces showing RTO and RTT evolution.*

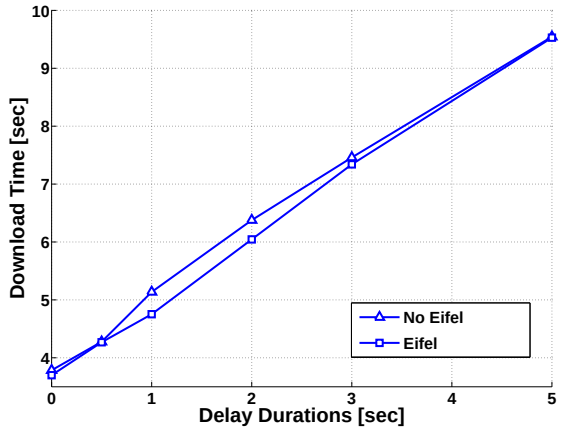
RTO quickly drops at the beginning of a connection. Hence, a small delay spike is able to trigger a retransmission when the delay is applied earlier. On the other hand, as more and more packets are being transmitted, the queueing delay gradually increases the RTT and TCP increases its RTO value. Thus, a larger delay spike is required to trigger a timeout when the delay occurs later during the connection. However, when timeouts occur later during the connection, the congestion window size is large, and more packets will be retransmitted than the case of earlier timeouts. Hence, even though the possibility of a timeout is reduced, the absolute retransmission ratio increases with the delay location.

Figure 6.11(a) shows the download times for delay spikes of different durations averaged over the different delay locations. It can be seen that the use of Eifel leads to up to 0.4 seconds improvement in the download time (which corresponds to a 9% decrease in the download time or a 13 Kbps increase in the goodput). When the delay duration is very low, Eifel performs almost the same as the non Eifel case because the number of retransmissions is small. For intermediate delay values, Eifel performs better. However, the performance degrades as the delay duration increases since the probability of multiple timeouts occurring increases, which disables Eifel's congestion window reversal [26].

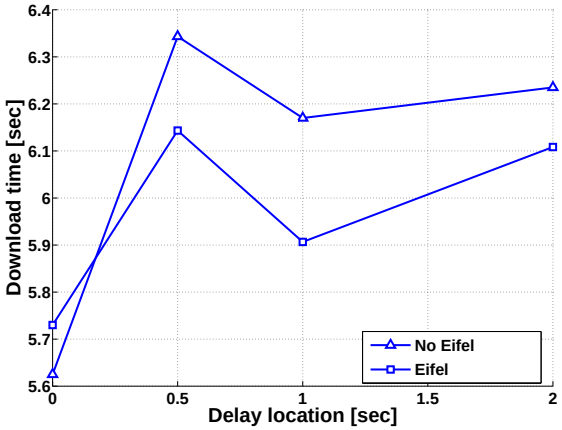
Figure 6.11(b) shows the download times for delay spikes applied at different locations averaged over the different delay durations. The download time is the highest for all the considered cases when the delays are applied at 0.5 seconds. As mentioned earlier, for lower delay location (with the exception of delay location of 0 sec), the possibility of a timeout occurring is very high, even for low delay durations, and hence the increase in the download time.

## 6.5 Summary

In this chapter the impact of spurious retransmissions on the performance of FTP file download sessions over a UMTS network has been investigated. The




*(a)Impact of delay duration*




*(b)Impact of delay location*

**Figure 6.11:** *Impact of delay duration and location: Mean download times.*

causes of spurious retransmissions considered were reordering caused by using RLC out of sequence delivery and delay spikes caused by many factors such as bandwidth downgrades or link outages that are masked through persistent link layer retransmissions.

Eifel is very effective in mitigating the problem of reordering caused by the RLC out of sequence delivery option, increasing the goodput by up to 33%. The use of SACK, however, causes performance degradation whether used with Eifel or not. This is because there is a higher probability of entering the fast retransmission phase earlier when using SACK as the sender does not have

to wait for triple duplicate ACK. Using DSACK also improves performance but the improvement in the goodput is not more than 14%.

From the delay spikes investigations it is found that although the use of Eifel improves performance, the improvement is not as noticeable as in the reordering cases (less than 9% improvement in the goodput). The reason for this is that higher delay values trigger multiple retransmissions, in which case the Eifel algorithm acts conservatively in the same way as in a normal slow start. Delays applied at different locations also have different impacts on performance. Lower delay values during the earlier stages of the FTP session are able to cause timeouts, while higher delay values are required when the delays are applied later.

The results show that for background services such as FTP, the use of in-sequence delivery gives optimal results. However, out-of-sequence delivery outperforms in-sequence delivery in terms of RTT and using Eifel lessens the goodput degradation of out-of-sequence delivery. Thus, we recommend the use of out-of-sequence delivery along with Eifel for interactive services such as web browsing where both the goodput and the RTT are important performance metrics.

# Part IV

# Heterogeneous Networks

# Chapter 7

# Packet Service Performance in Heterogeneous Networks

Several wireless technologies exist, each one with its own advantages and limitations. For example, GPRS/UMTS provide high mobility with limited bandwidth while WLAN provides higher bandwidth with restricted mobility. Thus, the current trend in mobile communications is not one network technology replacing another, but rather the inter-operability between different networks [101; 102]. Thus, the investigation of packet service performance in UMTS will not be complete without considering scenarios where UMTS coexists with other technologies, which is the rule rather than the exception [33; 103; 104].

In this chapter, the performance of packet services in UMTS when UMTS coexists with GPRS and WLAN networks is investigated. As in the previous chapter, the focus is on TCP and some of its extensions. The chapter is organised as follows. Section 7.1 gives an overview of heterogenous networks and the main issues that impact performance in such networks. The investigated cases are described in Section 7.2. Section 7.3 gives the investigation results. Finally, Section 7.4 summarises the main findings from the investigations.

## 7.1 Heterogeneous Networks

A heterogenous network is a networking environment that provides a user with a possibility of accessing services via different network technologies. Heterogeneous networks are also known as *overlay networks* [102; 105]. A typical example of an overlay network is where a user accesses services via a WLAN while inside an office and via a cellular network such as UMTS outside the office.

Figure 7.1 illustrates one such scenario. The user is initially accessing services using a WLAN network in the office and then leaves the office and the connection is handed to a UMTS network. This handover between two different network technologies is known as *vertical handover*, and specifically a handover from a network with a small coverage (and usually higher capacity) to another with a larger coverage (and usually lower capacity) is known as *upward vertical handover* [102]. On the other hand, a handover from a network with a large coverage to a smaller one is known as *downward verti-*

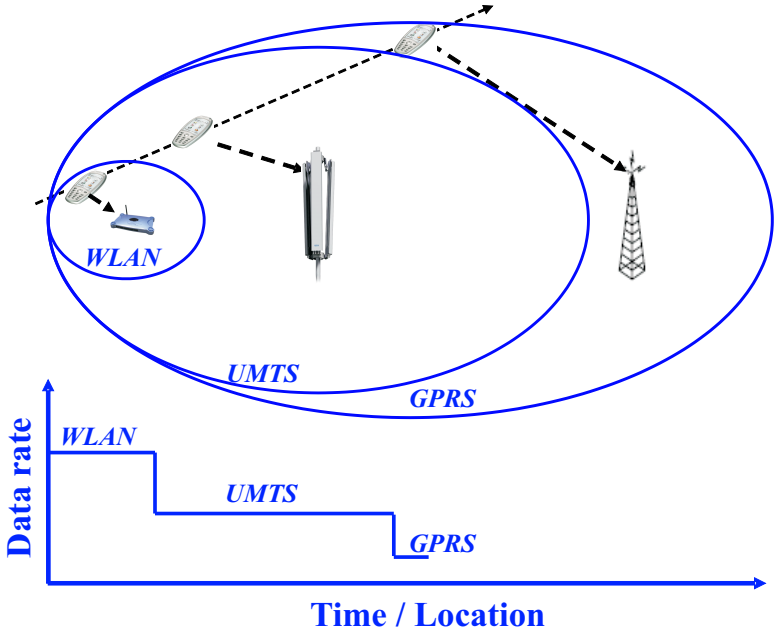



**Figure 7.1:** *An example showing a user accessing services in a heterogeneous network.*

*cal handover*. After a while, the user leaves the coverage area of the UMTS network and the connection is handed over to a GPRS network. A simple handover decision algorithm is one that lets the user get connected to the lowest overlay network, i.e the one with highest capacity, as long as it is within the range of the user, the network is not overloaded and the user's mobility is not leading to frequent handovers [102]. However, handover algorithms should also consider several other factors such as service cost, security and power consumption [106].

Vertical handovers can also be classified further as *hard* and *soft* [105]. In hard vertical handover, the user is connected to only one network at a time, i.e. the connection with the previous network has to be broken before the new one can be used. In soft handover, simultaneous connections with the concerned networks can be made and data can be sent/received simultaneously to/from the two network access points (see [103] and [104] for the issues concerning soft vertical handover implementation).

When it comes to handling mobility and associated vertical handover, several proposals exist that operate at the network (e.g. Mobile IP[107]), transport (e.g. using Stream Control Transmission Protocol (SCTP) [31]) or even application layer (e.g. using Session Initiation Protocol (SIP) [108]). The focus of this chapter is on the impact of vertical handover on TCP performance rather than the mechanisms for implementing handover. The reader is referred to [105; 109] and the references therein for the details of different handover management techniques.

### 7.1.1 Performance Issues and Suggested Solutions

When a vertical handover occurs[1], the data and ACK packets that are already in flight will be lost, unless some buffering mechanism is used to tunnel the data from the old connection to the new one when the handover is finished. This data loss will trigger TCP retransmissions and possibly slow start that

[1]From here onwards, unless otherwise specified, by vertical handover it is meant hard vertical handover.

will degrade the performance of the connection. Not only that, if the handover period is long enough, multiple timeouts will occur, and the RTO is doubled each time due to the exponential back-off behaviour described in Section 3.2.3. This might lead to a long idle period after the handover is finished where TCP waits for the RTO to expire.

Another effect of vertical handover that can lead to performance degradation is the change in the BDP [110; 111]. As described in Section 3.1.2, a window size equal to the BDP is ideal for a TCP connection because that guarantees full link utilisation. If a handover is made from a large BDP network to a small BDP network, even assuming ideal handover with no delay or loss, congestion will occur as TCP will keep on using the larger window size until congestion forces it to reduce it. On the other hand, a handover from a small BDP to a large BDP network will not take advantage of the large BDP and leads to lower utilisation for some time. Over-buffering (i.e. all transmission queues are set to the largest possible BDP the connection can be made through) is suggested to combat the impact of changing BDP [110; 111]. However, it is hard to find out the right buffer size beforehand, and also it might lead to congestion after handover as mentioned before [111].

If the UE[2] is equipped with a multi-mode interface that can simultaneously transmit/receive using multiple networks technologies, soft handover can be performed, basically eliminating the handover loss and delay. However, other handover associated problems arise in this case. If a handover is being made from a low bandwidth network to a higher one, reordering can result as the new packets are received/transmitted through the new interface, which will lead to performance degradation as described in the previous chapter. Similarly, ACK reordering can also occur, where ACKs for higher sequence numbers sent over the new link arrive before ACKs for older sequence numbers that are sent over the old link. This ACK reordering will lead to a more bursty transmission if the sender is increasing its window based on the amount of bytes ACKed, instead of just the number of ACKs received [111].

[2] Although mobility is not part of the definition of a UE as given in [45], in this report the UE is assumed to be mobile.

In [112; 113] the problem of slow start due to lost packets during handover and the impact of exponential back-off were addressed. In order to mitigate packet loss, the solution suggested was to buffer data that is arriving in the old access points and tunnel it to the new access point. This solution requires a high buffering capability at the base stations if they are serving several users. Also it was suggested that the UE sends a number of duplicate ACKs once the handover is over to trigger fast retransmissions and thereby lessen the impact of exponential back-off. However, this solution requires a communication between the link and transport layer in the UE in order to notify the transport layer when the handover is finalised.

Solutions that are based on splitting (or proxying) the TCP connection between the UE and the server at the access point exist and the most representative of these is I-TCP[3] [114]. In I-TCP, when a UE wishes to communicate with some fixed server, a request is sent to the current access point to open a TCP connection with the server on behalf of the UE. The UE communicates with its access point on a separate connection using a variation of TCP that is tuned for wireless links and is also aware of mobility. The UE TCP only sees an image of its peer TCP that in fact resides on the access point. It is this image that is handed over to the new access point in case the UE is handed over to another cell or network. The main drawback of I-TCP is that end-to-end semantics of TCP are not preserved, i.e. the access point sends ACKs to the server when it gets data, disregarding the fact that the packet might get lost during the transition between the access point and the UE. Also, long and frequent handovers demand large buffering requirements at the access point [115].

[3] As mentioned briefly in Chapter 5, split connection proposals are also applicable to standalone wireless networks as they can mask wireless channel errors.

## 7.2 Investigated Scenarios

### 7.2.1 Handover Buffering Mechanisms

Most of the methods mentioned in the previous section rely on buffering data during handover[4]. Thus, we focus our study to a general case where the concern is mainly on different buffering mechanisms and their impact on TCP performance.

Three different buffering mechanisms are considered:

- **No Buffering (NB):** All packets that were in flight when the handover is started and also the ones that arrive during handover are dropped.
- **Full Buffering (FB):** All packets that were in flight when the handover is started and also those that arrive during handover are queued. Most existing buffering schemes for vertical handover use full buffering.
- **Intermediate Buffering (IB):** This is a buffering scheme that we propose where all packets that were in flight when the handover is started are dropped but the ones that arrive during handover are queued.

We use Figure 7.2 to illustrate the three buffering mechanisms. For simplicity, only data packets are shown. First the UE is accessing the packets from the old access point, and since the wireless link has lower bandwidth a queue will develop at the access point. Immediately after the arrival of packet 8, a vertical handover is initiated, at which point there are four packets (packets 5–8) in the access point's buffer. Two more packets (9 & 10) arrive at the access point during handover, and after a while there is a retransmission (packet R) as the server is not getting any ACKs.

[4]There are also methods like M-TCP ([116]) and Freeze-TCP ([117]) that employ techniques to make the server aware of the handover and pause the transmission. However they require a prediction of handover and/or need a communication framework between the UE's link and TCP layers.

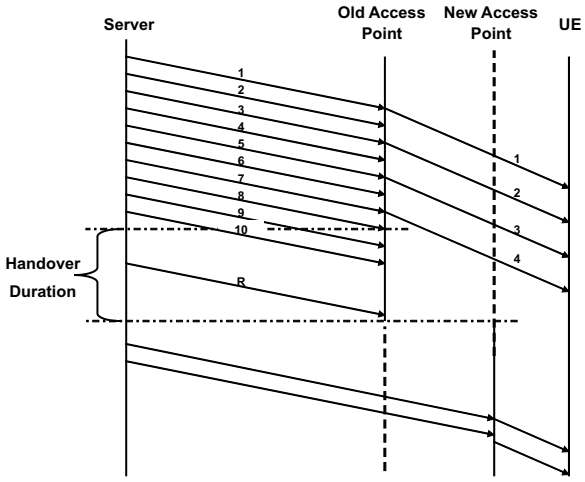



**Figure 7.2:** *Figure used to illustrate the different buffering mechanisms.*

If NB is employed, packets (5–10) will be flushed as well as the retransmitted packet. With FB, immediately after handover is finished, packets 5–10 as well as the retransmitted packet are forwarded to the new access point and then to the UE. With IB, packets 5–8 will be flushed while packets 9, 10 and the retransmitted packet are stored in the old access point and forwarded to the new access point and subsequently to the UE immediately after handover is finished.

For the IB and FB cases we assume that there is a handover protocol/ mechanism that enables the forwarding between the access points. IB is easier to implement (and requires less memory) than FB because the access point has to buffer the packets only after a handover is initiated, while in the FB case the non ACKed packets have to be buffered even before handover is initiated after they are transmitted to the UE.

### 7.2.2 Emulation Tool

As in the previous chapters, the RESPECT emulator is used. However considerable modifications have been made to the emulator for the handover studies (see Appendix B for the details). The main modification is that the network

behaviour is described as a state machine (with states representing different networks and sub-states defining the state within each network) instead of the implementation of the protocol stack in the original RESPECT.

Each sub-state of a network is characterised by a given bandwidth and latency. When a packet arrives at the emulator, if no handover is going on, the delay the packet is going to experience ($d_i$) is calculated as:

$$d_i = (\frac{s_i}{BW_i}) + L_i + q_i \tag{7.1}$$

where $s_i$ is the size of the packet, $BW_i$ is the currently allocated bandwidth when packet $i$ arrives, $L_i$ is the link latency (propagation delay + processing delay), and $q_i$ is the queueing delay which depends on the number packets that are currently on transit. A timer is started for each packet which has this value and the concerned packet leaves the emulator when the timer expires.

A time limit or amount of data transmitted can be used to specify when a transition should be made between networks or within the sub-states of a given network. Also, the three different buffering mechanisms mentioned above are implemented. No packet leaves the emulator during the handover period. When the handover is over, if there are any queued packets (i.e if IB or FB were used), the new delay that the packets are going to experience will be updated using (7.1) as if each queued packet is arriving at that instant. Note that this ensures that no reordering will occur when a handover is made from a slow link to a faster one, and we assume that no soft vertical handover is going on, i.e. data can not be sent using the two networks simultaneously.

### 7.2.3 Parameter Settings

Two different heterogenous network scenarios are investigated, one comprised of UMTS and WLAN, and another comprised of UMTS and GPRS. Both upward and downward vertical handover are investigated for the two scenarios. A 1 Mbyte file is downloaded, and handover is initiated immediately after half

of the file has been downloaded. These values for the file size and the handover initiation time are used to ensure that TCP has reached its steady state when the handover starts. The link layers of the concerned mobile networks are assumed to employ a reliability mechanism to mitigate wireless channel errors, i.e. packet losses, if any, are only due handover. Different TCP extensions which are available on the Linux platform are tested, for the same reason as in the previous chapter. Table 7.1 summarises the main parameters.

**Table 7.1:** *Summary of main parameter settings.*

| Parameter | Value(s) |
|---|---|
| UMTS Bit rates (DL/UL) (Kbps) | 384/32 |
| UMTS one way latency (ms) | 75 |
| GPRS Bit rates (DL/UL) (Kbps) | 32/32 |
| GPRS one way latency (ms) | 350 |
| WLAN Bit rates (DL/UL) (Kbps) | 1000/64 |
| WLAN one way latency (ms) | 5 |
| File sizes (Mbyte) | 1 |
| Eifel (Timestamps) | on/off |
| TCP SACK | on/off |
| TCP DSACK | on/off |
| Handover delay duration (sec) | [0, 2, 5, 10] |

## 7.3 Emulation Results

In this section, the results from the emulation runs are presented. First, the results from WLAN to UMTS upward handover case are presented. Particular results from the downward handover case and handover between UMTS and GPRS are presented later on.

### 7.3.1 WLAN to UMTS Upward Handover

#### 7.3.1.1 Ordinary TCP

Figure 7.3 shows the download times when a file download that started during a WLAN connection is handed over to a UMTS connection midway through

the download session. Linux TCP with SACK and Eifel disabled was used for the file downloads.

Notice that even when there is no handover delay, all the packets in flight are dropped with NB and IB. This, combined with the fact that no data will arrive during handover as the handover delay is zero, is the reason why the two buffering mechanisms perform exactly the same when there is no handover delay. FB in this case results in a 2 seconds decrease in the download time (from 17.8 to 15.8 seconds), which translates to a 13% (60 Kbps) improvement in the average goodput (from 471 Kbps to 531 Kbps).

When the handover delay is increased to 2 seconds, the use of IB leads to a slight improvement in the download time (from 18.5 to 18 seconds, which translates to a 13 Kbps or 3% improvement in terms of goodput) while FB leads to performance degradation, increasing the download time by 0.6 seconds, and this is equivalent to a 14 Kbps or 3% decrease in the goodput. The trend is similar for the 5 seconds delay case. This seems strange at first look, but the TCP sequence traces reveal the reason behind this result.

Figures 7.4 - 7.6 show the TCP sender's behaviour during handover for the

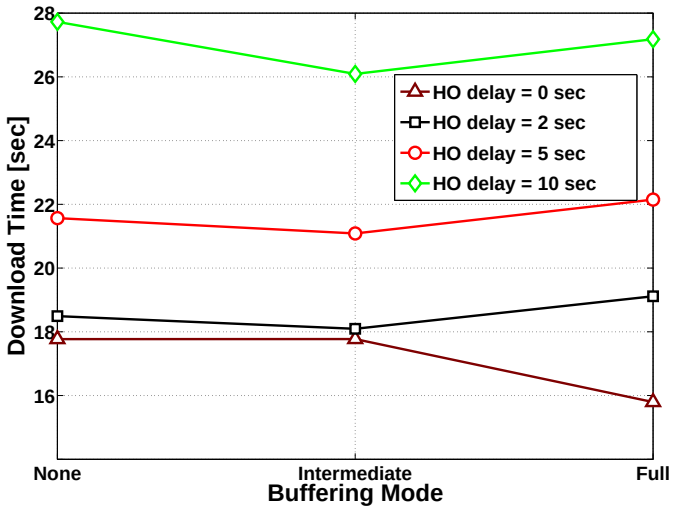



**Figure 7.3:** *Mean download times: WLAN to UMTS handover, SACK and Eifel disabled.*

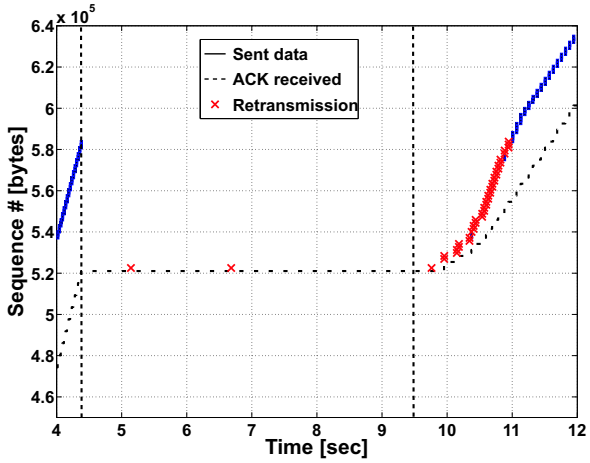



**Figure 7.4:** *Time sequence graph: WLAN to UMTS handover, 5 seconds handover delay, SACK and Eifel disabled, No buffering.*

three different buffering schemes when the handover delay is 5 seconds. The vertical dotted lines in the figures indicate the start and end of the handover period.

Figure 7.4 shows that during handover, at 5.14 seconds, there is a timeout (the RTO at this time was 770 ms) and the concerned packet is retransmitted. A second and third timeout occur at 6.7 and 9.7 seconds, owing to TCP's

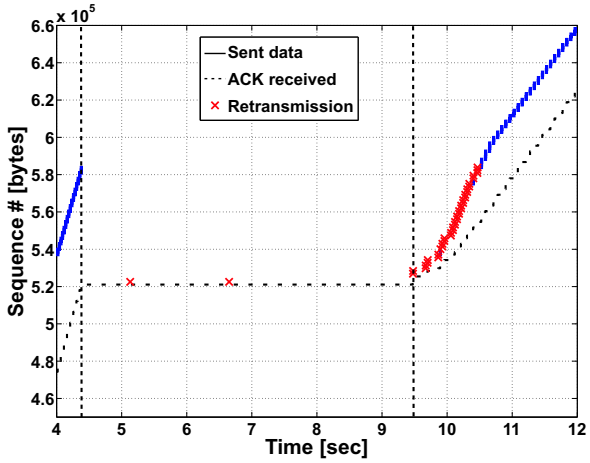



**Figure 7.5:** *Time sequence graph: WLAN to UMTS handover, 5 seconds handover delay, SACK and Eifel disabled, Intermediate buffering.*

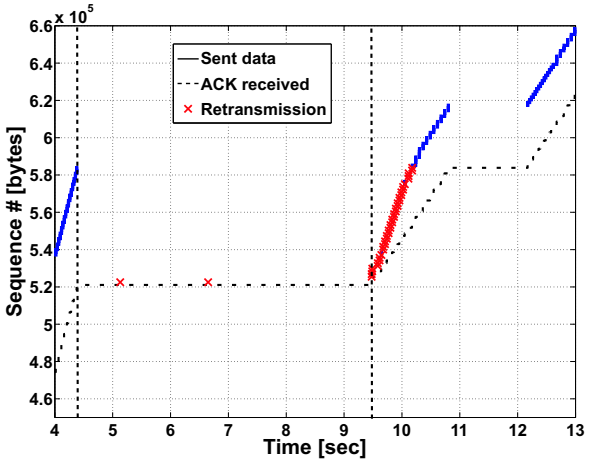



**Figure 7.6:** *Time sequence graph: WLAN to UMTS handover, 5 seconds handover delay, SACK and Eifel disabled, Full buffering.*

exponential back-off mechanism. The first two retransmissions are lost since nothing is buffered. The receiver will get only the third retransmission, and when the ACK for this packet arrives, the rest of the packets that were lost are retransmitted (as can be seen in the figure between around 10 to 11 seconds).

Figure 7.5 is similar to Figure 7.4 during the handover period, but as the retransmissions are buffered, the receiver will get them immediately after the handover, and it responds with an ACK. Thus, IB saves the sender from the impact of exponential back-off, which was the cause of the wait after the handover is finished for NB.

In Figure 7.6, the impact of exponential back-off is also reduced but this time, not only the retransmission is buffered but also the whole window of data that was lost at the start of handover for the other two buffering modes. When the receiver starts getting the buffered data, it sends ACKs and the sender assumes these ACKs correspond to the retransmissions and keeps sending the rest of the window. After that, the retransmitted packets actually start arriving at the receiver, and the receiver responds by sending duplicate ACKs. Fortunately, thanks to the new-reno modification of TCP congestion control, these duplicate ACKs will not drive the sender into fast retransmission. However, the sender will have to stay idle for about 1.4 seconds while waiting for a new

ACK to advance its window, and hence the performance degradation due to FB.

From Figure 7.3 it can also be seen that as the handover delay increases, the degradation due to FB decreases and it improves the performance as compared with NB. This is because a larger handover delay increases the impact of the exponential back-off as the RTO is doubled for each retransmission. Thus, with NB, the idle time between the end of the handover and the reception of the next retransmission (which can be seen in Figure 7.4) also increases, worsening its performance. As the exponential back-off does not affect IB and FB, any kind of buffering leads to improvement for large handover delays, but still IB performs better (6% improvement in goodput for the 10 seconds handover) than FB (2% improvement).

#### 7.3.1.2 Eifel

Figure 7.7 shows the download times when Eifel is enabled. Comparing the values from this figure with that of Figure 7.3 shows that for handover delay values of 0 and FB, the performance is the same regardless of the Eifel setting, while for NB and IB, the performance worsens with Eifel (additional 1.1 seconds in the download time).

Figure 7.8 shows the time sequence graph for the case of Eifel used along with NB when the handover delay is zero. As can be seen from the figure, after the handover is finished (there is only one vertical bar because the handover delay is zero), there is a timeout and a retransmission at around 5.2 seconds. An ACK is received after a small delay[5] and this ACK contains a timestamp earlier than the retransmitted packet. This is because the original transmission of the retransmitted packet was received properly, but the corresponding ACK was dropped due to handover. Thus, Eifel finds out that particular retransmission was erroneous, reverts the congestion window and transmits two new packets. The reception of these new packets generates only a duplicate

[5]Linux employs delayed ACKs (except for the first few packets) and one ACK is sent for every two packet received or after the reception of one packet if nothing is received after certain delay.

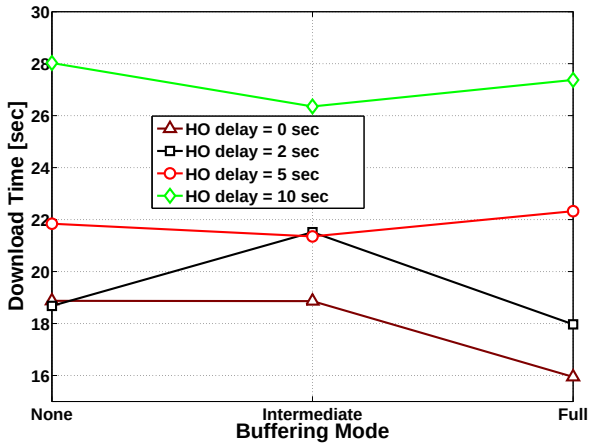



**Figure 7.7:** *Mean download times: WLAN to UMTS handover, Eifel enabled, SACK disabled.*

ACK. Although the decision that the retransmission of that particular packet was spurious is right, it leads to performance degradation. The sender has to wait for another timeout (at around 7 seconds), thus becoming idle for 1.3 seconds before it starts retransmitting the packets that are actually lost. Please refer to Appendix E for a detailed analysis of this behaviour.

Comparing Figures 7.3 and 7.7 again, it can be seen that when the handover

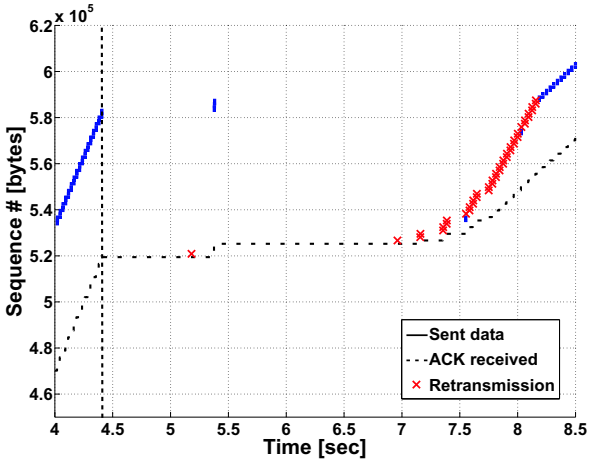



**Figure 7.8:** *Time sequence graph: WLAN to UMTS handover, no handover delay, Eifel enabled, No buffering.*

delay is increased to 2 seconds, the result is the same for NB, but using Eifel leads to performance degradation with IB (a 3.4 seconds increase in the download time, equivalent to 16% or 73 Kbps decrease in the goodput). For FB, Eifel leads to a 1.2 seconds improvement in the download time, corresponding to 6% or 26 Kbps increase in the goodput. Figures 7.9 - 7.11 illustrate this behaviour.

As can be seen from Figure 7.9, when NB is used, there is a timeout during handover and since nothing is buffered, there is another timeout after handover, and after that the rest of the window is retransmitted. The behaviour is different from that of Figure 7.8 because, as discussed in the previous chapter, Eifel becomes more conservative and stops reverting the congestion window when multiple timeouts occur for a given packet [26].

In Figure 7.10 a similar effect to that of Figure 7.8 can be seen, because multiple timeouts are avoided due to the buffered retransmission. Thus, Eifel finds that particular retransmission was spurious and starts sending new data. As in Figure 7.8, another timeout is required to detect the loss of the other packets. The performance degradation is clearly visible as only 600000 bytes are sent by 11 seconds in this case, while more than 700000 bytes are sent out at the same time for NB.

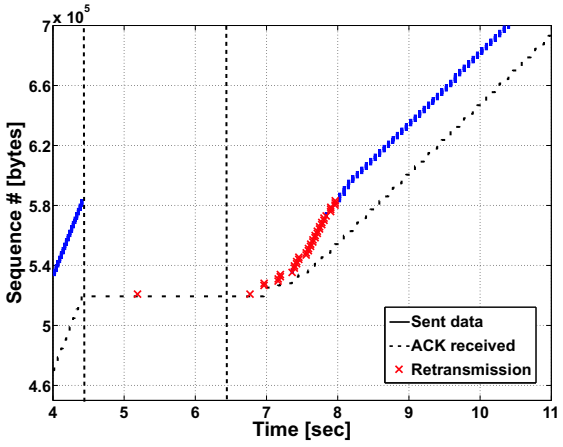



**Figure 7.9:** *Time sequence graph: WLAN to UMTS handover, 2 seconds handover delay, Eifel enabled, No buffering.*

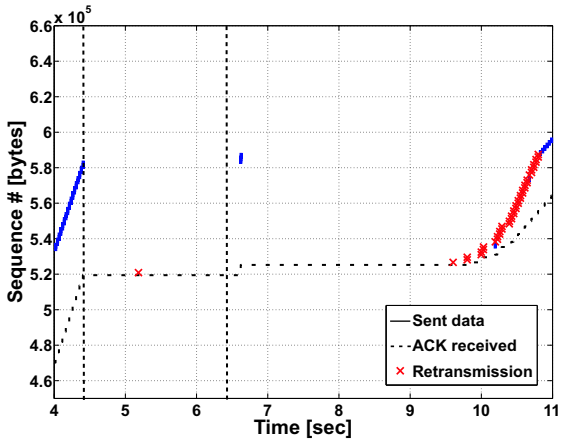



**Figure 7.10:** *Time sequence graph: WLAN to UMTS handover, 2 seconds handover delay, Eifel enabled, Intermediate buffering.*

In Figure 7.11, there is only one retransmission, which is buffered during handover and sent out immediately after. As there was only one timeout, the Eifel algorithm is in full operation. It detects the retransmission was spurious and reverts the congestion windows back to the state just before the handover, and transmission of new data is resumed. Since all the packets were queued, this decision will lead to performance improvement as compared with the previous case. The sender is able to send around 800000 bytes by 11 seconds.

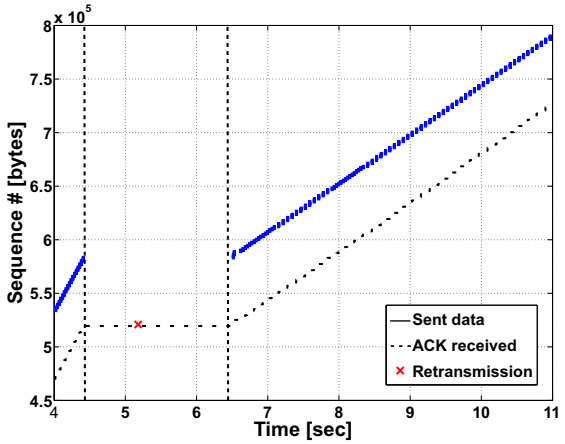



**Figure 7.11:** *Time sequence graph: WLAN to UMTS handover, 2 seconds handover delay, Eifel enabled, Full buffering.*

When the handover delay is increased to 5 or more seconds, there are multiple timeouts regardless of the buffering mode, and the Eifel algorithm becomes as conservative as ordinary TCP. Thus, the results shown in Figures 7.3 and 7.7 for these cases are the same.

#### 7.3.1.3 SACK

Figure 7.12 shows the download times when SACK is enabled. Comparing this with Figure 7.3, the only difference is when there is no handover delay and FB is not used. For this case, the performance degradation, as compared with ordinary TCP and no handover delay, is 5.3 seconds in the download time, equivalent to a 30% (108 Kbps) reduction in the goodput. The reason for the similarity for the cases when there is a handover delay is because for those cases, there will definitely be a timeout. And when there is a timeout, slow start is activated as in normal TCP [58; 54].

Figure 7.13 shows a time sequence graph illustrating the performance degradation due to TCP SACK usage when there is no handover delay. As can be seen in the figure, after the handover, the sender gets a delayed ACK from the

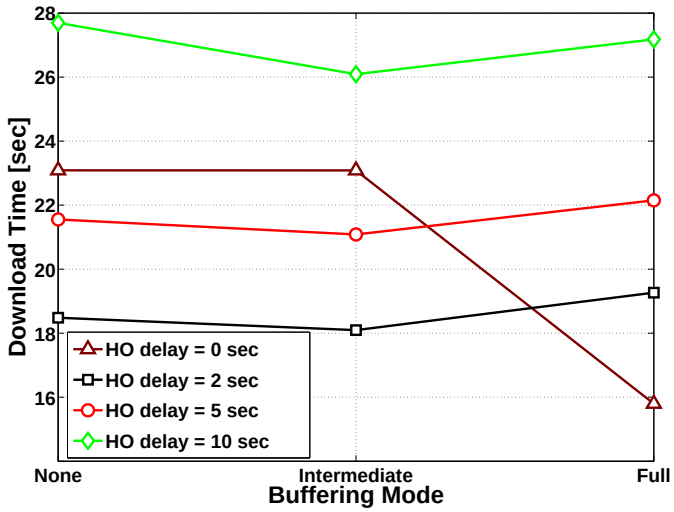



**Figure 7.12:** *Mean download times: WLAN to UMTS handover, SACK enabled, Eifel disabled.*

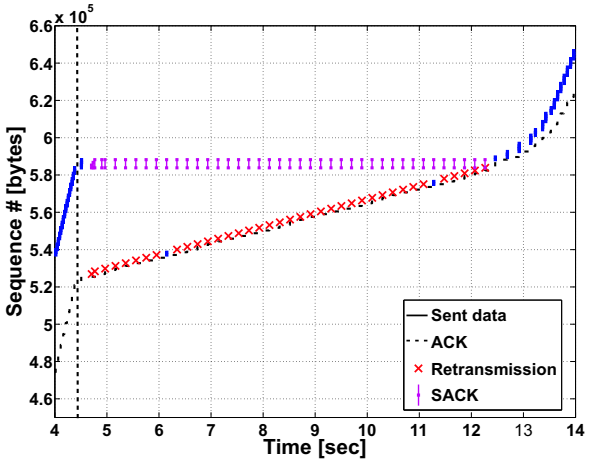



**Figure 7.13:** *Time sequence graph: WLAN to UMTS handover, no handover delay, SACK enabled, No buffering.*

receiver. This delayed ACK results in the retransmission of two new packets. When the receiver gets these packets, it responds by sending duplicate ACKs with SACK information. As the SACKed block is several packets above the packet acknowledged by the duplicate ACK, TCP SACK immediately retransmits the concerned packet. This particular step is advantageous as compared with the no SACK case, because there is no waiting for a timeout or a triple duplicate ACK. However, due to the conservative SACK based recovery employed by Linux [58; 54], the TCP sender will retransmit one packet at a time when an ACK with a SACK block arrives. It takes more than 7 seconds for the sender to recover from the packet losses. If SACK was not used, the sender would have waited for a timeout. However, after that the retransmission of all the lost data would have taken a shorter time because it would have been done in slow start mode where two retransmissions are sent for every incoming ACK.

The results when SACK is enabled along with Eifel follow the same pattern as the non-Eifel case, i.e. the zero handover delay case leads to performance degradation while the other cases perform the same.

#### 7.3.1.4 DSACK

The results when DSACK is enabled show the same behaviour as in SACK because when the DSACK blocks arrive it is already too late, and the bandwidth is already wasted through unnecessary retransmissions. An example illustrating this is shown in Figure 7.14. Comparing this with Figure 7.6, it can be seen that the only difference is that the receiver sends a DSACK block whenever it gets a duplicate segment, i.e. when the unnecessary retransmissions are received. Unfortunately, the sender is not able to make proper use of this information because all the unnecessary retransmissions have already been sent out.

### 7.3.2 Other Results

#### 7.3.2.1 UMTS to WLAN Downward Handover

The upward and downward handover results are basically the same except for these three notable exceptions. First, for the 2 seconds and 10 seconds handover delays, FB performs better than NB when ordinary TCP is used. The

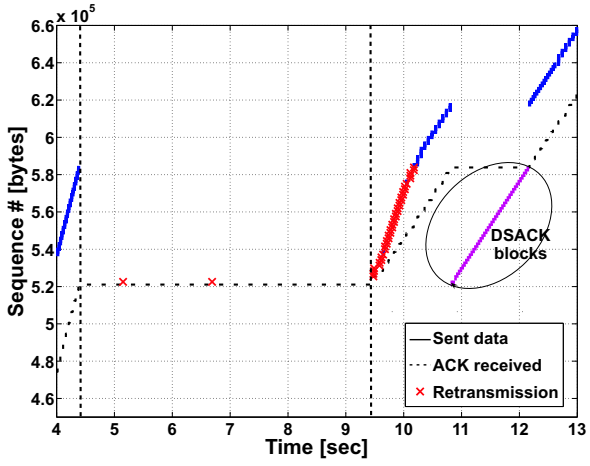



**Figure 7.14:** *Time sequence graph: WLAN to UMTS handover, 5 seconds handover delay, DSACK enabled, Full buffering.*

RTT of the packets just before handover in this case is around 1.4 seconds while it was 0.52 seconds for the upward handover case. The corresponding RTO value is around 1.8 seconds. Thus, after the first timeout, for the 2 seconds delay and NB case, additional 3.4 seconds have to elapse before there is another timeout and the lost data can be retransmitted[6]. Redundant retransmission is performed as before for the FB case, but still it takes less than the 3.4 seconds that the sender has to wait for the NB case. The 10 seconds handover delay case follows a similar pattern.

Secondly, the SACK effect discussed in Section 7.3.1.3 for the zero handover case leads to performance improvement instead (a 1.1 seconds decrease in the download time, equivalent to a 6% (30 Kbps) increase in the goodput as compared with the no SACK case) because the retransmissions are sent over a WLAN link that has a smaller RTT and higher bandwidth. Figure 7.15 shows the time sequence plot illustrating this. Comparing this with Figure 7.13, the recovery took only 2.4 seconds, which is one third of the time it took for the upward handover case (roughly equal to the ratio between the data rates of the two networks).

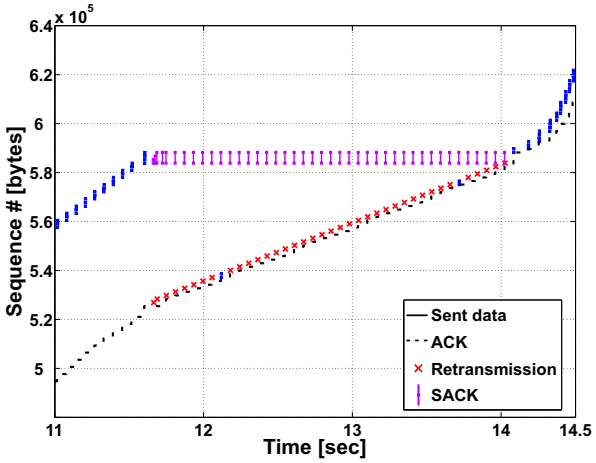



**Figure 7.15:** *Time sequence graph: UMTS to WLAN handover, no handover delay, SACK enabled, No buffering.*

[6]The first timeout will occur after 1.8 seconds, and due to exponential back off the second timeout will occur 3.6 seconds after the first timeout, i.e. 5.4 seconds from the start of the handover. The handover is finished 2 - 1.8 = 0.2 seconds after this second timeout period is started. Thus the sender remains idle for 3.6 - 0.2 = 3.4 seconds.

The third major change that is noticed between the upward and downward handover cases is when Eifel is used. This is because during a handover delay of 5 seconds, there will be only one timeout since a second timeout will require 5.4 seconds. This means that the Eifel algorithm will be fully operational for IB and, making the same decision as described in section 7.3.1.2, will lead to performance degradation. On the other hand, FB gives the best performance for all handover cases except the 10 seconds handover.

#### 7.3.2.2 Handover between UMTS and GPRS

The results for the UMTS to GPRS upward handover follow the same trend as that of the upward handover between WLAN and GPRS, i.e. IB gives the best performance for most of the cases. However, the results were quite different for the downward handover between GPRS and UMTS.

Figure 7.16 shows the results for this case. As can be seen from the figure, IB degrades performance for all the cases except when there is no handover delay, increasing the download time by up to 20 seconds for the 10 seconds handover delay case (equivalent to 12% (5 Kbps) decrease in goodput). Fig-

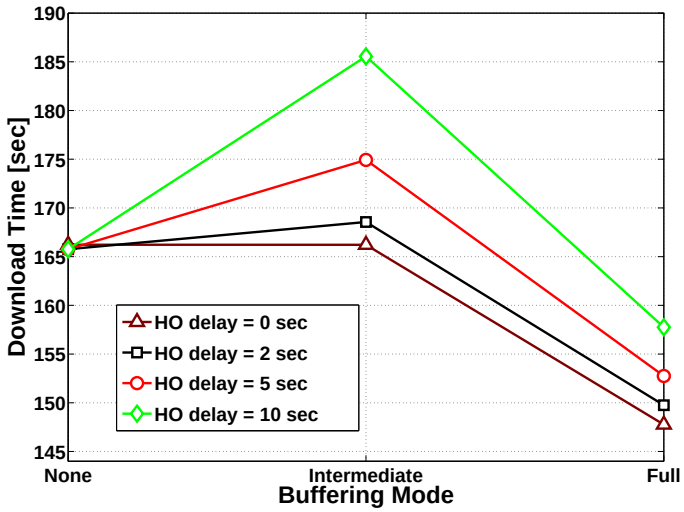



**Figure 7.16:** *Mean download times: GPRS to UMTS handover, Eifel and SACK disabled.*

ures 7.17 and 7.18 show the traces for the case of 10 seconds handover delay for the NB and IB, respectively, to illustrate the differences.

In Figure 7.17, a timeout occurs around 8 seconds after the handover is finished, and the retransmission of the lost packets resumes. On the other hand, in Figure 7.18, immediately after handover, the buffered ACKs that were saved during handover arrive at the sender and they cause the retransmission of new data. Since an extra 10 seconds was introduced in the RTT of the packets that just got ACKed, the RTO is also increased to reflect this change. Hence, the sender waits an extra 17 seconds after handover before starting retransmissions.

Returning to Figure 7.16, it can also be seen that when NB is used the results are the same regardless of the four different handover delay cases considered. Also FB performs better for all the considered cases. This is due to the RTT value when handover starts. The RTT values for the case of 10 seconds handover delay for the different buffering schemes is shown in Figure 7.19. When the handover is started the RTT is more than 15 seconds, which is well over the maximum handover delay considered, i.e. only one timeout will occur because of handover for NB and IB and there will be none for FB. The figure also shows the increase in the RTT due to buffering immediately after handover for IB and FB.

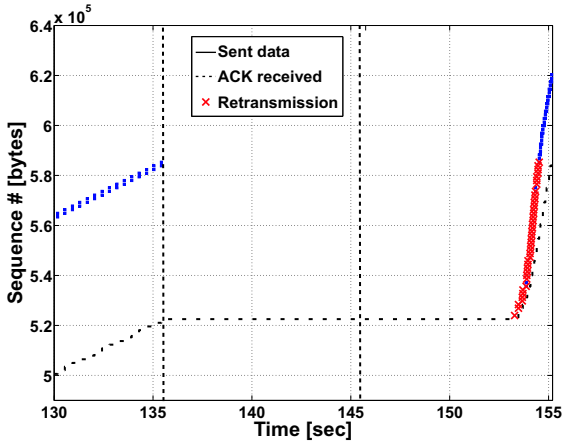



**Figure 7.17:** *Time sequence graph: GPRS to UMTS handover, 10 seconds handover delay, Eifel and SACK disabled, No buffering.*

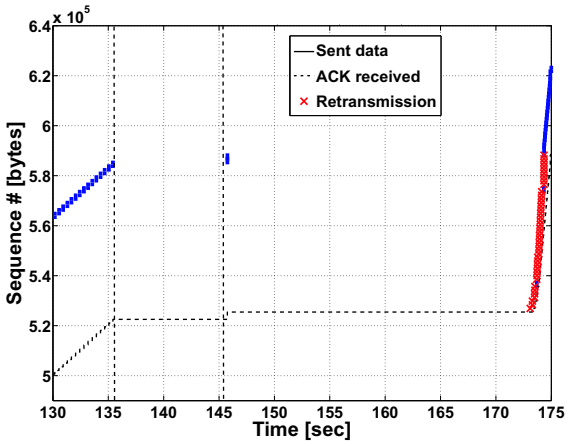



**Figure 7.18:** *Time sequence graph: GPRS to UMTS handover, 10 seconds handover delay, Eifel and SACK disabled, Intermediate buffering.*

Figure 7.20 shows the download times for the GPRS to UMTS handover case when Eifel is enabled. Comparing this with Figure 7.16 it can be seen that the use of Eifel has an impact for non zero handover delays when NB is used, while the rest of the cases are almost the same as before.

When there is no handover delay, immediately after handover a delayed ACK is received, which properly acknowledged everything the receiver has got. So when a timeout occurs for the cases of NB and IB, it will be for an

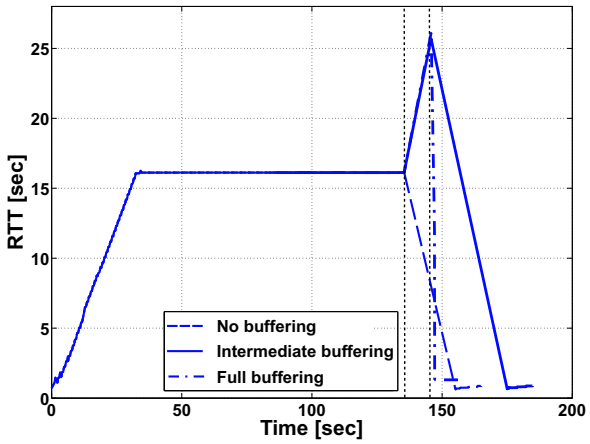



**Figure 7.19:** *RTT progression: GPRS to UMTS handover, 10 seconds handover delay, Eifel and SACK disabled.*

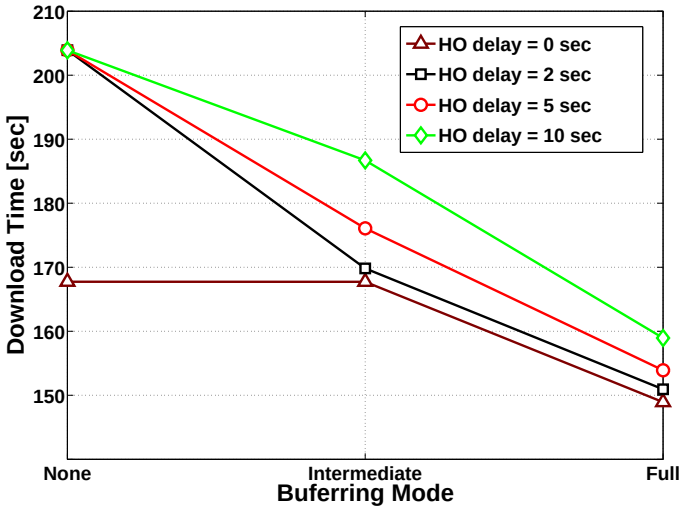



**Figure 7.20:** *Mean download times, GPRS to UMTS handover, Eifel enabled, SACK disabled.*

actually lost packet (instead of the spurious retransmission due to lost ACKs described in Section 7.3.1.2). Thus, Eifel will not detect a spurious retransmission, and hence will not affect performance.

For all the IB cases, the effect will be similar as the delayed ACK will be buffered during handover too. For FB, there are no retransmissions, thus Eifel has no impact.

When there is a handover delay and NB is employed, Eifel degrades the performance considerably (increasing the download time by 38 seconds, which corresponds to a 23% (9 Kbps) decrease in the goodput). This is because the delayed ACK that was sent by the receiver a short while after handover is started is lost, and timestamps will behave in the same way as described in Section 7.3.1.2 leading to performance degradation.

Finally, Figure 7.21 shows the download times for the GPRS to UMTS handover case when SACK is enabled. Comparing this with Figure 7.16 it can be seen that the use of SACK has no impact for FB as there are no retransmissions.

As described previously while explaining the Eifel results, for NB with no

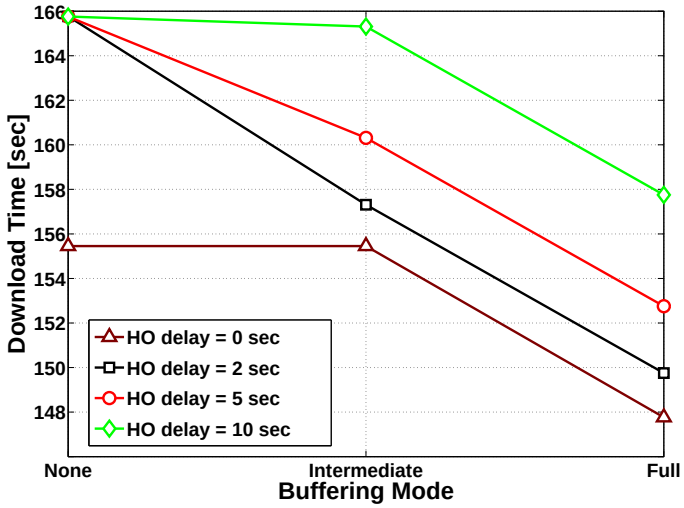



**Figure 7.21:** *Mean download times, GPRS to UMTS handover, SACK enabled.*

handover delay and for all the IB cases, a delayed ACK will be received by the sender, which responds by sending new packets. The reception of these new packets will cause the generation of SACK blocks, which results in immediate retransmission (in a similar fashion to that shown in Figure 7.13). This avoids the waiting for the timeout, which was around 20 seconds. Thus, even though only one packet is sent per round trip time due to SACK recovery, this is still much faster than waiting for a timeout and using slow start for the retransmissions. The performance improvement is up to 20 seconds in download time or 11% (5.5 Kbps) as compared with the no SACK case.

When NB is employed and the handover delay is not zero, the performance remains the same as in Figure 7.16. This is because the delayed ACK will be lost and no new data will be sent. Thus, no SACK blocks will be sent by the receiver and the sender has to wait for a timeout before retransmitting the lost packets.

## 7.4 Summary

In this chapter, the performance of packet services in a heterogenous network comprised of UMTS, WLAN and GPRS has been investigated. The behaviour

of ordinary TCP, Eifel and SACK under different handover conditions has been studied. A new buffering scheme called Intermediate buffering (IB), where only packets that arrive during handover are buffered, is proposed. The performance of IB is compared with Full buffering (FB), where all the packets in-flight when the handover is started are buffered in addition to the packets that arrive during the handover, and also with No buffering (NB) schemes. Table 7.2 summarises the results for all the cases considered in this chapter. In the table, the comparisons are made row wise. The signs "+ +", "+", "-" and "- -" refer to the best, better (more than average), worse (less than average) and worst performers for the concerned handover case, respectively.

**Table 7.2:** *Summary of performance results for the different handover cases, buffering schemes and TCP extensions considered.*

| **No Handover Delay** | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| | **No Buffering** | | | **Intermediate Buffering** | | | **Full Buffering** | | |
| | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK |
| **WLAN-UMTS** | + | - | - - | + | - | - - | + + | + + | + + |
| **UMTS-WLAN** | - | - - | + | - | - - | + | + + | + + | + + |
| **UMTS-GPRS** | + | - | - - | + | - | - - | + + | + + | + + |
| **GPRS-UMTS** | - | - - | + | - | - - | + | + + | + + | + + |
| **Small Handover Delay (2 seconds)** | | | | | | | | | |
| | **No Buffering** | | | **Intermediate Buffering** | | | **Full Buffering** | | |
| | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK |
| **WLAN-UMTS** | + | + | + | + + | - - | + + | - | + + | - |
| **UMTS-WLAN** | - | - | - | + + | - - | + + | + | + + | + |
| **UMTS-GPRS** | + | + | + | + + | - | + + | - - | + + | - - |
| **GPRS-UMTS** | - | - - | - | - | - | + | + + | + + | + + |
| **Medium Handover Delay (5 seconds)** | | | | | | | | | |
| | **No Buffering** | | | **Intermediate Buffering** | | | **Full Buffering** | | |
| | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK |
| **WLAN-UMTS** | + | - | + | + + | + + | + + | - - | - - | - - |
| **UMTS-WLAN** | + | + | + | + + | - - | + + | + | + + | + |
| **UMTS-GPRS** | + + | + | + + | + + | + | + + | - - | - | - - |
| **GPRS-UMTS** | + | - - | + | - | - | + | + + | + + | + + |
| **Large Handover Delay (10 seconds)** | | | | | | | | | |
| | **No Buffering** | | | **Intermediate Buffering** | | | **Full Buffering** | | |
| | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK | Ordinary | Eifel | SACK |
| **WLAN-UMTS** | - | - - | - | + + | + + | + + | - | - | - |
| **UMTS-WLAN** | - | - - | - | + + | + + | + + | + | + | + |
| **UMTS-GPRS** | + | + | + | + + | + | + + | - - | - | - - |
| **GPRS-UMTS** | + | - - | + | - | - | + | + + | + + | + + |

For the no handover delay cases, FB performs the best. For non zero handover delay cases and ordinary TCP, IB is the best performer. The only exception is the GPRS to UMTS downward handover, where a delayed ACK that was buffered during handover made TCP to overestimate the RTT, and TCP timeout took a longer time. FB performs even worse than NB for the upward handover cases because the whole window of lost data is retransmitted even though it was buffered. For downward handover cases, FB performs better than NB as the impact of exponential back-off is more severe due to the larger RTT in the first (slower) network in which the file download is started from.

Eifel degrades the performance of IB for the cases where a single timeout occurs. This is because there is actual data loss but Eifel detects a spurious retransmission when a delayed ACK that was buffered during handover arrives at the sender immediately after handover and the sender has to wait for another timeout to detect that actual data loss has occurred. For NB, the performance also becomes worse with Eifel for most of the cases because a delayed ACK is lost instead and there will be a retransmission later on. Eifel detects this retransmission is spurious when an ACK is received, and thus the sender has to wait for yet another timeout to retransmit the lost data. Eifel improves the performance of FB because there is no actual data loss in this case, and Eifel detects spurious retransmissions, if any, and avoid unnecessary retransmission of the whole window of data and also reverts the congestion window.

SACK has no effect on the performance of FB as there is no data loss, and hence no SACK blocks are generated. For IB, SACK has a positive impact on the performance in the downward handover from GPRS to UMTS because the lower RTT in UMTS makes the one packet per RTT recovery employed by SACK to be more effective than waiting for a timeout with the RTO of a GPRS network. There is no change in performance due to SACK in the upward handover case between UMTS and GPRS (and also for both vertical and downward handover cases between UMTS and WLAN) because a timeout occurs and slow start is activated instead of SACK based recovery. For NB, if there is a handover delay, everything is lost, so there is also no SACK blocks generated. However, when there is no handover delay, SACK has a positive impact for the downward handover cases and a negative impact for the upward

case, for the same reasons as in IB.

# Part V

# Subjective Quality

# Chapter 8

# Subjective Evaluation of Packet Service Performance

QoS is usually identified by some basic performance metrics such as delay, throughput and jitter. In the previous chapters, the performance investigations were carried out based on these objective measures. As the main impact of QoS provision is on the end user, a detailed study on QoS should involve the end user. So in this chapter, a different approach is taken where the performance is evaluated using subjective measures that are acquired through usability testing. The purpose of this testing is twofold: to find out if there is any noticeable trend in users' perceived QoS and if so, if the subjective measures are in line with objective measures of quality.

The term *usability testing* is broadly defined in [118] as *"a process that employs participants who are representative of the target population to evaluate the degree to which a product meets specific usability criteria"*. The usability criteria is a combination of many factors such as learnability, efficiency, ease of use, error proneness, and overall satisfaction [119]. In this chapter, the only usability criteria of interest is the general user satisfaction as we are not concerned about new services but only about the network through which they are provided.

The chapter is organised as follows. Section 8.1 describes the usability test

design and setup. The results from the experiments are analysed in Section 8.2. Finally, Section 8.3 summarises the findings from this chapter.

## 8.1 Usability Test Design and Setup

The main concern while designing a usability test is to make sure that the test is *reliable* and *valid* [119; 120]. *Reliability* addresses the issue whether the results will be similar if the test is to be repeated several times. On the other hand, *validity* is concerned with whether the result actually reflects the usability issues that are being tested.

Usability tests can be made more reliable by making sure that the conditions under which the tests are carried out are similar for each test user. However, due to the inherent differences (in terms of experience, taste, mood, etc...) between the test users, assuring reliability is not always so easy. Reliability can be improved by testing as many users as possible and analysing the results using standard statistical methods such as Analysis of Variance (ANOVA)[1].

Validity is a more subtle issue because it is mainly dependent on the test methodology and not only the variability of test users. The major validity problems arise due to:

- choice of the wrong group of users, for example, users to which the product or service may not be relevant to in real life
- choosing the wrong tasks, for example, asking users to deal with very simplified tasks as compared to real tasks
- confounding effects, for example, inadvertently changing a parameter that is not being tested along with a parameter that is being tested

A usability test has to be designed properly before being carried out to

[1] ANOVA is a statistical method that can be used to check if there is a significant difference between the means of multiple sets of data by splitting the overall observed variance into within-conditions and between-conditions variances [73; 121].

ensure that it is as valid and reliable as possible. A proper test design should specify the *purpose* of the experiment, *user profiles*, *test methodology*, *test environment*, *task lists*, and *evaluation measures* [118; 120]. In the following sections each of these facets of the usability test design process for the experiments carried out in this chapter are described.

### 8.1.1 Purpose

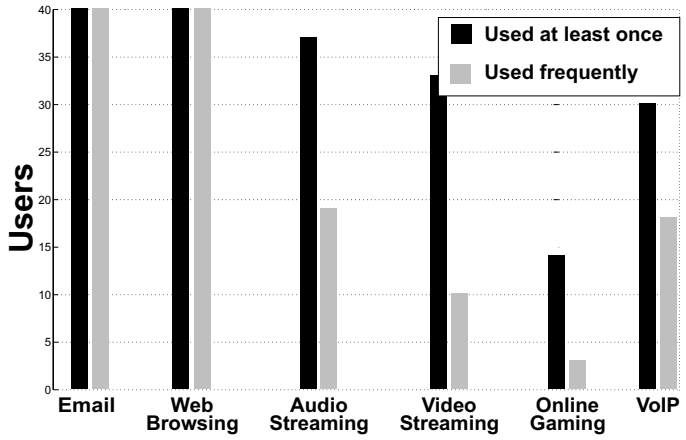
The overall goal of the usability experiment conducted here is to find out the effect of network characteristics on users' perception of quality of web browsing and streaming services in mobile networks, and to see if the users' perception matches with objective quality measures. Web browsing and streaming are chosen not only because they are widely used on the Internet, but also because they are already available in many recent mobile devices. For web browsing, the considered environment is a UMTS network, and the test investigates the impact of FER, bandwidth, and RTT. For the streaming services, the environment is a heterogeneous network comprised of UMTS and WLAN, and the test investigates the impact of handover-induced bandwidth upgrades and downgrades.

The results of this evaluation can be useful for operators, service providers, network equipment manufactures and researchers in the telecom area. The findings will help to indicate which factors have a major impact on the perceived QoS when it comes to web browsing and streaming services. Based on the results, the concerned stakeholders will be able to optimise their network to bring more satisfaction to the end user and as a consequence increase their revenue.

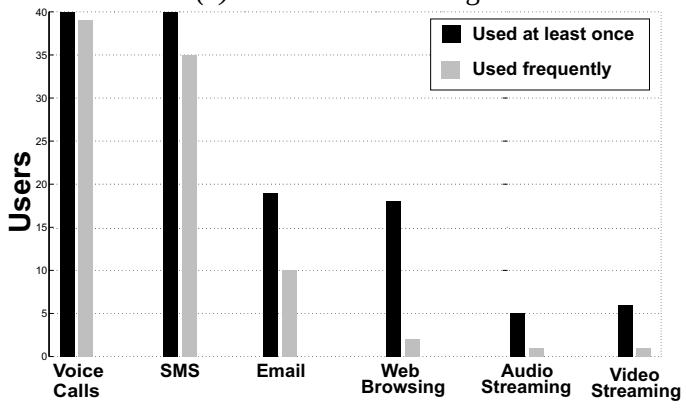
### 8.1.2 User Profile

According to [119] at least 30 users are required to get acceptable statistics for a usability experiment if the results are gathered through questionnaires. Thus, a total of 40 participants, of which 8 are female, are tested during the

first two weeks of April, 2006 at Aalborg University. Due to time and resource limitations most of the participants are colleagues and friends from and around the university. All of the participants are above the age of 23, and ten are above the age of 40. All the participants are mobile phone users. Figures 8.1(a) and 8.1(b) show the experience of users with regard to services on the wired Internet and via their mobile devices, respectively. All the users have access to high speed fixed Internet (10 Mbps+) connection at work/school, and they used the Internet on a daily basis. All users but one also have a fixed Internet connection of 500 Kbps+ at home.




*(a)* Internet service usage




*(b)* Mobile service usage

**Figure 8.1:** *Users' service experience.*

### 8.1.3 Methodology

Usability tests can be conducted using either *independent groups* or *within-subjects* design [118; 120]. In independent groups design, subjects are assigned to different experimental conditions, while in within-subjects design (also known as *repeated measures*) all subjects experience all the experimental conditions.

Within-subjects design is preferable when there is a shortage of test users, but it has the disadvantage that the results may be biased if there is any possibility of learning during the experiment which can affect the conditions to be tested later on during a given session. Also, if the time required for testing is substantial, independent grouping is preferred, as user tiredness can influence the end result.

Within-subjects design is used in the usability tests conducted here because of the limited number of subjects available. There is a minimal risk of learnability in the usability tests conducted here, because all the users are already aware of the services under investigation. There is, however, a risk of bias depending on the order of the conditions being tested. For example, if the first network condition being tested is the best one, the user's expectation can become very high and the latter conditions might end up being rated very low. This bias was mitigated by using randomised testing, where the different test conditions were presented in a random fashion to each user. The test session is also designed to last only 40 minutes in order to avoid the bias due to tiredness, albeit this means a reduction in the number of conditions to be tested.

The different test sessions have to be conducted in a consistent manner, i.e. keep all the conditions apart from the ones being tested as constant as possible. In order to insure consistency, all the test sessions are moderated by one test conductor and written instructions are provided to the subjects as well as oral ones to avoid the risk of leaving out some details. At the beginning of each session, each subject is given an orientation to the experiment being conducted which clearly specifies the purpose of the experiment and what is expected from them. The instructions are read off from a script (which is shown in

Appendix D.1) in order to make sure each subject gets the same information. The instruction basically tells the subjects the purpose of the experiment is to test the effect of different network conditions on perceived QoS and that they have to assume that they are accessing the services through a mobile network.

### 8.1.4 Test Environment

The test environment has to be as realistic as possible. Also, the environment should be consistent from one subject to another. Taking these two factors (i.e. consistency and realism) into account, the test environment is setup as shown in Figure 8.2. As can be seen from the figure, the browsing and streaming services are provided through a local server instead of the Internet. This ensures that results from the different users will not be affected by external factors such as network congestion, i.e. ensure consistency. Internet Information Server (IIS)$^{TM}$ from Microsoft Cooperation and Helix$^{TM}$ streaming server from Real Networks are used for hosting the web pages and the streaming videos, respectively. Internet Explorer$^{TM}$ with disabled caching is used for browsing, while Real Player$^{TM}$ is used as the streaming client application.

The services are accessed through a Fujitsu-Siemens$^{TM}$ LT P-600 tablet PC connected via a stable 54 Mbps WLAN to a network emulator. The WLAN link to the emulator is used only to make the setup more realistic as the subjects are told that they have to assume that the services were accessed via a mobile

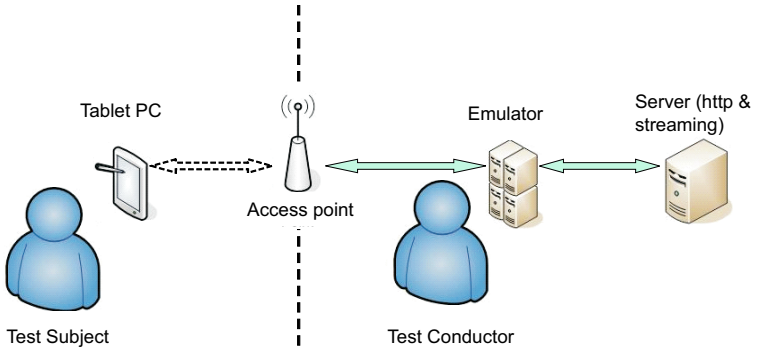



**Figure 8.2:** *Usability test setup.*

network. The subject under test is the only one connected via the access point to ensure that the WLAN link will not end up being a bottleneck.

The original RESPECT emulator as well as the modified version that supports heterogeneous networks (see Appendix B) are used to emulate different network conditions.

The test conductor's role is mainly to change the configuration files used by the emulator for the different settings to be tested. The actual selection of a configuration is done via an automation script in a randomised fashion, making both the test subject and test conductor completely blind to the conditions being investigated.

Table 8.1 shows the different settings that are tested for the web browsing studies. The different cases are numbered like $Bandwidth\_RTT\_FER$. An initial setup time of 1 second is included to emulate the setup time of a DCH for all the cases in Table 8.1.

For the heterogeneous network settings, shown in Table 8.2, the test is started with "Network 1" as the initial network and after 30 seconds a handover is initiated, which takes 5 seconds, and the connection is then handed over to "Network 2". Network 2 is kept till the end of the test case. Also, no packets are lost during handover. The RTT for UMTS is set at 150$ms$, while for WLAN it is set to 10$ms$. The cases are numbered like $Bandwidth\_WLAN$ or $WLAN\_bandwidth$, depending on whether the streaming is started on WLAN or UMTS, and "bandwidth" is the bandwidth of the UMTS connection. The

**Table 8.1:** *UMTS network setup.*

| Case# | Bit rate ($Kbps$) | RTT ($ms$) | FER(%) |
|---|---|---|---|
| 384_150_1 | 384 | 150 | 1 |
| 384_150_10 | 384 | 150 | 10 |
| 384_150_20 | 384 | 150 | 20 |
| 384_600_1 | 384 | 600 | 1 |
| 32_150_1 | 32 | 150 | 1 |
| 32_150_10 | 32 | 150 | 10 |
| 32_150_20 | 32 | 150 | 20 |
| 3840_20_0 | 32 | 20 | 0 |

**Table 8.2:** *Heterogeneous network setup.*

| Case# | Network 1 (Bit rate in *Kbps*) | Network 2 (Bit rate in *Kbps*) |
|---|---|---|
| 384_WLAN | UMTS (384) | WLAN (1000) |
| WLAN_384 | WLAN (1000) | UMTS (384) |
| 128_WLAN | UMTS (128) | WLAN (1000) |
| WLAN_128 | WLAN (1000) | UMTS (128) |
| 32_WLAN | UMTS (32) | WLAN (1000) |
| WLAN_32 | WLAN (1000) | UMTS (32) |
| WLAN | WLAN (1000) | WLAN (1000) |

last settings, for both the UMTS and heterogeneous network setup, are reference cases to get baseline results.

### 8.1.5 Task List

The subjects are given three tasks to complete, and each task is repeated for the different test conditions. The first two tasks are concerned with web browsing in a UMTS network. In the first task, users are asked to browse a local version of the “www.google.com" page (2 Kbyte text + one 9 Kbyte image, see Figure 8.3(a)), which can be considered as one of the smallest pages that one can encounter on the Internet these days. The second task is concerned with browsing a local version of a content-rich web page, “www.nokia.dk/phones/models/" (54 images of total 204 Kbyte + 166 Kbyte text and Java script, see Figure 8.3(b)). For both cases, eight repetitions are made, i.e. all the network settings from Table 8.1 are considered.

The last task is concerned with the streaming of a movie trailer in a heterogeneous network[2]. Twenty five of the subjects watched the trailer of “X-Men III", a sci-fi thriller, while the rest watched the trailer of “Open Season", an animation. The different scenes in the trailer for X-Men III are very short, and we thought that the jump from scene to scene might be interpreted by some users as the manifestations of delay jitter in the connection. Thus, the Open Season trailer, which is mainly comprised of one continuous scene, is used

[2]The subjects were not told that the videos were streamed via a heterogeneous network. They were only told that the services are provided through a mobile network.


*(a)* Google page

*(b)* Part of the Nokia page

**Figure 8.3:** *Web pages used for the web browsing experiments.*

for control purposes. Both trailers are one and a half minutes long, and they are encoded into a SureStream$^{TM}$ file [122; 123] for 350 Kbps and 100 Kbps connections, so that an encoding rate that is appropriate for the connection can be chosen dynamically by the streaming client and server. The users watched the movie trailer seven times, each corresponding to a setting from Table 8.2.

In order for the users to be sure a given task is complete, an explicit Successful Completion Criteria (SCC) is required [118]. For the web browsing sessions, since the connection parameters are kept constant for a given test case, the subjects are told that they can stop the session any time if they have

already decided about the connection's quality. This reduces the duration of each experiment because for some of the cases, the web page download can take considerable time. For the streaming services, the SCC is defined as the time when the movie stream finishes playing.

### 8.1.6 Evaluation Measures

The subjective satisfaction is measured by a questionnaire that the subjects filled during the test. The International Telecommunication Union (ITU) recommends three main ways of determining perceived quality [124] [3]:

- **Absolute Category Rating (ACR):** The subject rates each test case one at a time based on a certain scale.
- **Degradation Category Rating (DCR)**: The (expected) best test case is shown to the user and the user is asked to compare and rate another case based on the best case.
- **Comparison Category Rating (CCR)**: This is similar to DCR but in this case, pairwise comparison is made in no particular order, i.e. the first test case the user experiences can be better or worse than the second one.

ACR leads to shorter test durations than DCR and CCR, as no pairwise comparison is involved. Also, it is less biased than DCR because the user is not aware whether conditions are improving or worsening beforehand. Due to these reasons, ACR is used for most of the test subjects. However, for the streaming tests, eight of the subjects performed a combination of ACR and DCR, where they are shown the trailer on the server machine (i.e. showing them the maximum quality they could expect based on the encoding) before they began rating the different cases. This is done as a control case, to see if

[3] The categories were originally defined for determining perceived acoustic quality of voice calls over a CS telephone network, but they are equally applicable here.

other factors such as original encoding quality and frame rate were biasing all the results.

Based on the recommendations in [124; 118], a scale from one (“Unacceptable") to five (“Excellent") is used to rate the quality of each repetition of the three different tasks. The questionnaire used is shown in Appendix D.2. The test conductor changes the settings and tells the subject to go ahead with the next task. The user performs the task and grades the case that was just tested. The process is repeated till all the cases are dealt with. Table 8.3 summarises the main parameters used in the usability experiments.

**Table 8.3:** *Main parameter settings for usability investigations.*

| **Parameter** | **Value(s)** |
|---|---|
| *Common Settings* | |
| # users | 40 |
| Services | Web browsing, Streaming |
| *Web browsing settings* | |
| Web pages | Google (2 Kbyte text + 9 Kbyte image (1 image file))<br>Nokia (166 Kbyte text + 204 Kbyte image (54 files) |
| Initial setup time (sec) | 1 |
| Bit rates (Kbps) | 384, 32 |
| RTT (ms) | 150, 600 |
| FER (%) | 1, 10, 20 |
| *Streaming settings* | |
| Video Clips (Trailers) | X-Men III, Open Season |
| RTT (ms) | 150 (UMTS), 10 (WLAN) |
| FER (%) | 0 |
| Clip duration (sec) | 90 |
| Encoding rate (Kbps) | 350 + 100 (SureStream) |
| Initial buffering (sec) | 10 |
| Handover cases | UMTS (384, 128, 32 Kbps) <–> WLAN (1000 Kbps) |
| Handover delay (sec) | 5 |

## 8.2 Results

### 8.2.1 Web Browsing

Figure 8.4 is a box plot that shows the distribution of the ACR results of all the eight different settings for the part of the test concerned with browsing (for

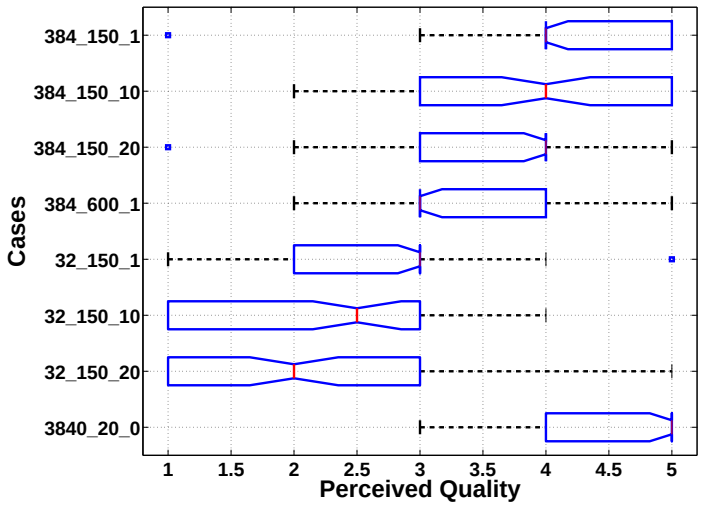


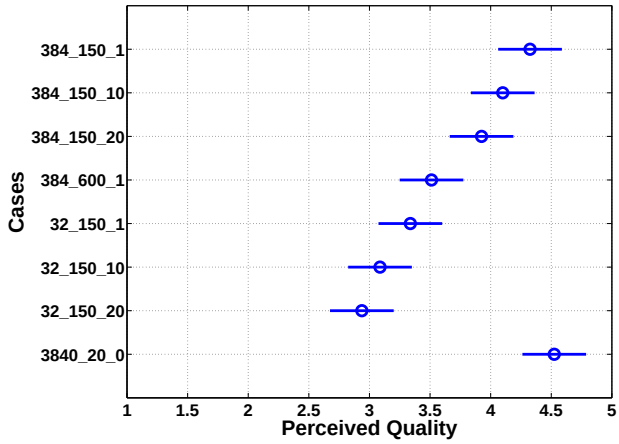

**Figure 8.4:** *Distribution of ratings for web browsing perceived quality.*

both the Google and Nokia pages). The boxes have lines at the lower quartile, median (the notches in the box), and upper quartile values. The whisker lines extending from each end of the boxes show the extent of the rest of the data set (the whiskers extend to $1.5\times$ the interquartile range), and outliers are those points beyond the ends of the whiskers. From the figure it is clear that there are differences between the median values for the different cases and only few outliers are present.

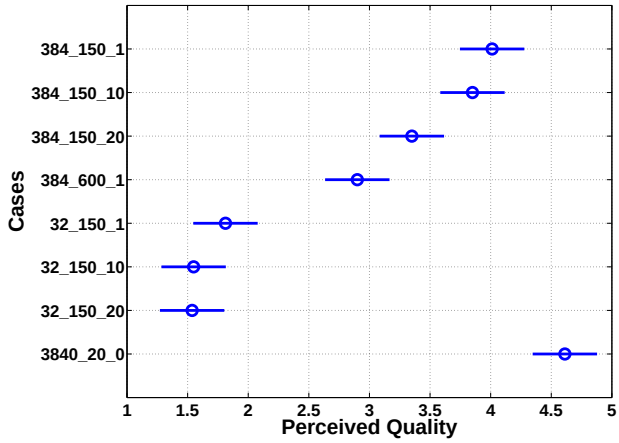
ANOVA and multiple comparisons are performed on the results to validate the differences in the means and the results are shown in Figures 8.5(a) and 8.5(b). In the figures, if two cases that are being compared have disjoint intervals, then they are significantly different from each other.

As expected, the differences between the different cases is more pronounced for the Nokia page (Figure 8.5(b)) than for the Google page (Figure 8.5(b)), due to the differences in the size of the two pages. However, even for such a small page as Google, there are quite noticeable differences.

The effect of FER and bandwidth on the ratings is shown in Figure 8.6. For the Google page, as can be seen from Figure 8.6, a change in the bandwidth from 384 Kbps to 32 Kbps lowers the rating by one grade for each considered




*(a)*Google page




*(b)*Nokia page

**Figure 8.5:** *Confidence intervals for the perceived quality of the web browsing experiments.*

FER. On the other hand, the Nokia page ratings are lowered by two grades for the same changes. For a given bandwidth, higher FER values lead to lower grades for both pages. However, the differences are not that significant for the Google page as can be seen from the overlap in the multiple comparison plots of Figure 8.5(a). For the Nokia page, the FER differences also do not lead to significant rating differences except between FER values of 1% and 20% for the 384 Kbps case.

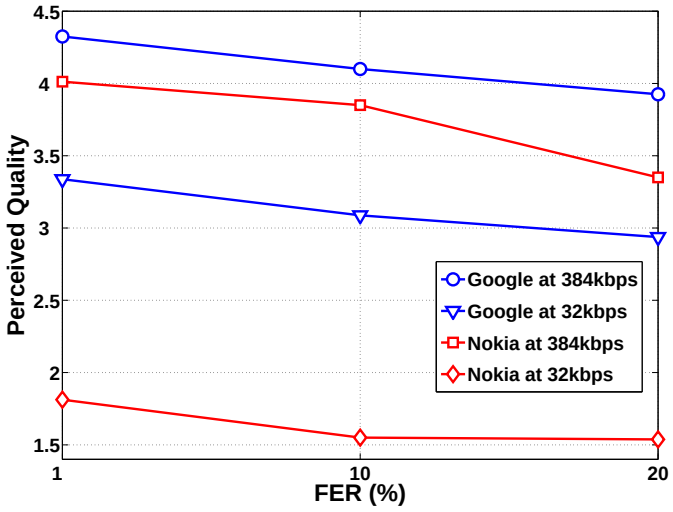



**Figure 8.6:** *Effect of FER and bandwidth on perceived QoS.*

Figure 8.7 shows the effect of RTT on the ratings, for a bandwidth of 384 Kbps and an FER of 1%. As can be seen from the figure, the RTT variation causes noticeable difference in the ratings for both pages. The reason for this is that even for the small Google page, there is a noticeable delay at the beginning for high RTT values.

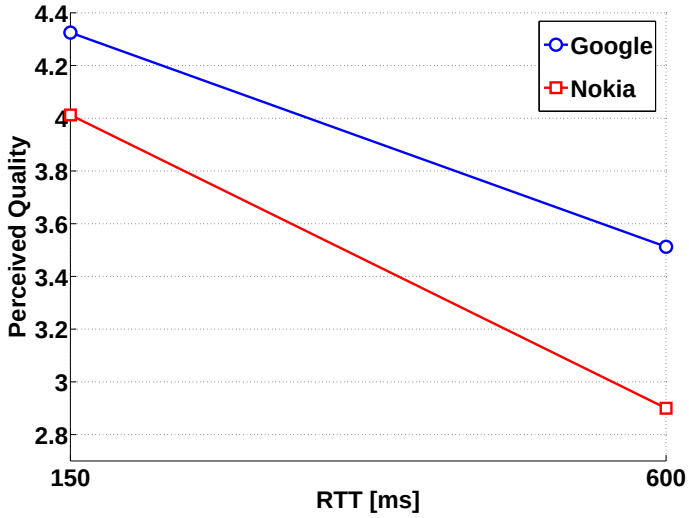



**Figure 8.7:** *Effect of RTT on perceived QoS.*

**Table 8.4:** *Initial and total delays vs perceived quality.*

| Case | Initial Delay (seconds) | Total Delay (seconds) | Perceived QoS |
|---|---|---|---|
| Google | | | |
| 3840_20_0 | 1.1 | 1.8 | 4.6 |
| 384_150_1 | 1.4 | 2.4 | 4.3 |
| 384_150_10 | 1.8 | 3 | 4.1 |
| 384_150_20 | 1.9 | 3.4 | 3.9 |
| **384_600_1** | **2.4** | **5.8** | **3.5** |
| **32_150_1** | **2** | **5.2** | **3.3** |
| 32_150_10 | 2.2 | 6.2 | 3.1 |
| 32_150_20 | 2.6 | 6.8 | 2.9 |
| Nokia | | | |
| 3840_20_0 | 4.6 | 13.6 | 4.6 |
| 384_150_1 | 6.6 | 19.2 | 4 |
| 384_150_10 | 8.8 | 24.2 | 3.9 |
| 384_150_20 | 10.4 | 28.6 | 3.4 |
| 384_600_1 | 15.4 | 49.4 | 2.9 |
| 32_150_1 | 38 | 99 | 1.8 |
| 32_150_10 | 45 | 121.8 | 1.6 |
| 32_150_20 | 53.8 | 144 | 1.5 |

A closer look at the results is done by using the objective measures of *initial* and *total* delay for the different cases, as shown in Table 8.4. Initial delay refers to the time before the skeleton of the page is downloaded (i.e. text and place holders for images), while total delay is the total download time. As can be seen from the table, the lower the initial delay and the total delay, the better the rating. The only exception are the *384_600_1* and *32_150_1* cases for the Google page.

Though the two cases are not significantly different from each other as can be seen from Figure 8.5(a), one would have expected the *32_150_1* case to be rated a bit higher than *384_600_1*, due to its lower initial and total delay values, but the reverse is true. It is found that for the *384_600_1* case, although the total and initial times are longer owing to the larger RTT, the Google logo comes in one go, while for the *384_150_1* case, the logo (which was not in interlaced format)[4] becomes visible gradually from top to bottom, owing to the slow speed of the connection. This small difference is the most probable

[4]Interlaced images appear on-screen by first displaying a lower resolution of the image and gradually showing the high resolution version.

explanation to the reversed ratings.

### 8.2.2 Streaming

As mentioned in Section 8.1.5, some of the users watched a different movie clip than the others. Analysis of the results showed that there are no significant differences between the ratings of the users who watched the different movies. Hence, all the results in this section refer to the combined ratings for all the forty users unless otherwise specified.

Figure 8.8 shows the distribution of the results of the seven different settings for the part of the test concerned with streaming. As in the case of section 8.2.1 there are very few outliers, and there seems to be a clear difference between the median values. ANOVA and multiple comparisons are made on the results to validate the differences in the means and the result is shown in Figure 8.9.

From the figures, it can be concluded that the $WLAN\_384$ and $WLAN$ cases are the highly rated ones, followed by the $384\_WLAN$ case. The $WLAN\_128$ case performs better than the $128\_WLAN$, $128\_WLAN$,and $WLAN\_128$ cases,

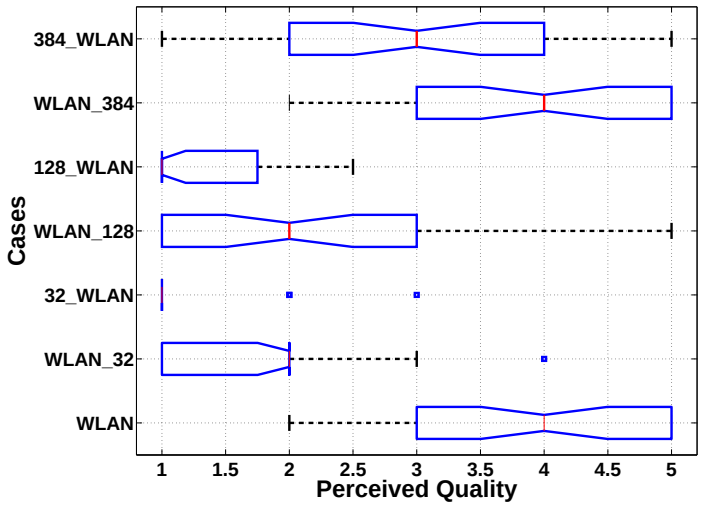



**Figure 8.8:** *Distribution of ratings for streaming perceived quality.*

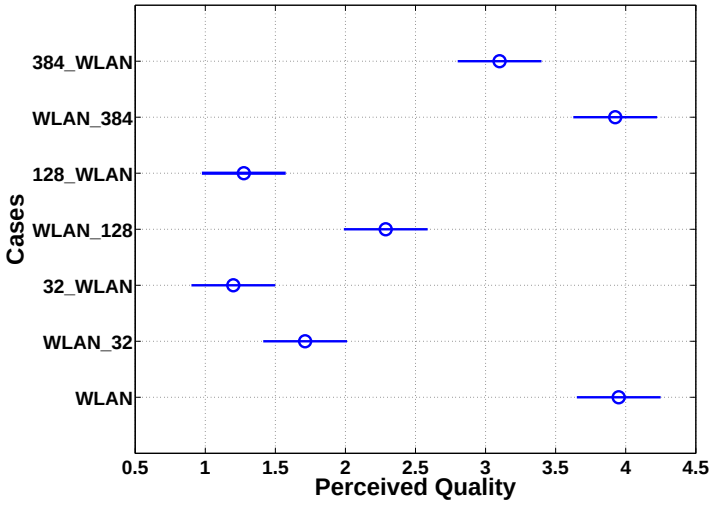



**Figure 8.9:** *Confidence intervals for the streaming video perceived quality.*

with no significant difference between the latter ones.

The 384_*WLAN* case is rated around one grade lower than the *WLAN*_384 case mainly due to the startup buffering time required. In the former case, it takes around 10 seconds before the first frames are shown, followed by around 10 seconds of buffering before the movie starts to be streamed continuously. In the latter case, the movie starts in around 10 seconds of buffering. Apart from this initial startup buffering differences, the two cases are the same because a 384 Kbps connection is able to support the maximum encoding rate of the stream. This is also the reason why the *WLAN*_384 and *WLAN* cases are rated the same.

A similar pattern is observed between the *WLAN*_128 and 128_*WLAN* cases, where the latter is rated a grade below the former. Apart from the startup buffering differences, the fact that 128 Kbps is not able to support the maximum encoding rate also plays a part to the observed quality of the two cases. For the *WLAN*_128 case, a good video quality is observed for the first half of the clip and then (after the handover), re-buffering is initiated. After the re-buffering is finalised, the streaming continues with the same video quality but it is soon interrupted again after 15 seconds for another re-buffering. After this

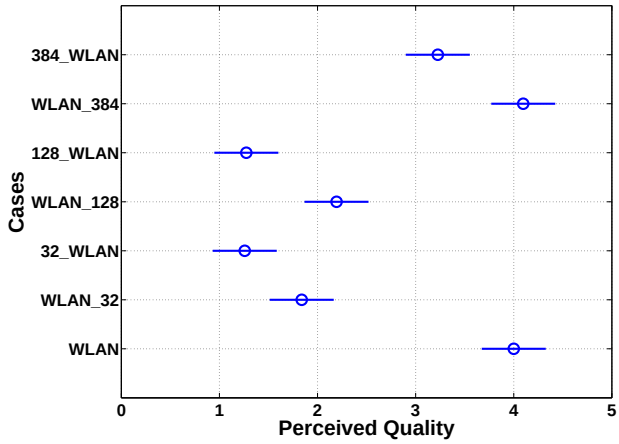
re-buffering, the clip is completed without interruption. According to [123], real player and Helix server are capable of detecting bit rate downgrade and stream the appropriate sub-stream from a SureStream file. However, for the *WLAN*_128 case, the duration of the movie was not long enough to see this at work.

For the 128_*WLAN* case, to begin with, it takes around 28 seconds before the movie starts and after 8 seconds of streaming at high quality, there is a very brief re-buffering and the stream continues with the downgraded bit rate of 100 Kbps, which is more of a slide show than a movie stream. It was not possible to get the description of the algorithms behind real player's and Helix server's adaptation to bit rate upgrade and downgrades, but from the results described here (detection in the 128_*WLAN* and no detection in case the *WLAN*_128), it can be concluded that they use a time weighted moving average of the bit rate to decide which sub-stream to use.

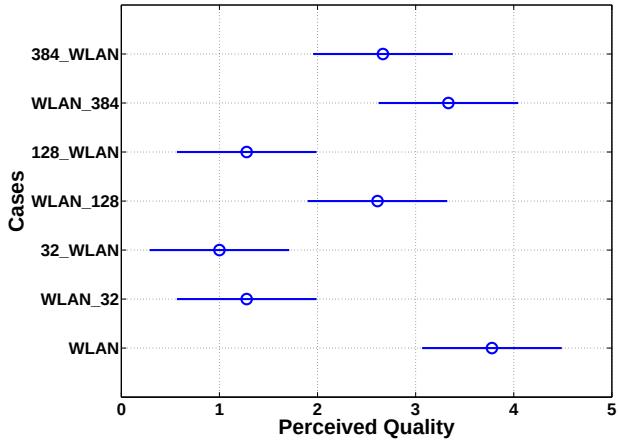
The *WLAN*_32 case is rated higher than the 128_*WLAN* case, though the differences are not significant. This small difference can be attributed not only to the initial startup buffering time, but also to the streaming encoding and bandwidth detection behaviour of real player and Helix server. For the *WLAN*_32 case, after the handover, the streaming bit rate is also not downgraded. However, due to the very low available bandwidth, a slide show effect similar to that of the 128_*WLAN* case is seen (albeit with a lower frame rate) immediately after the first re-buffering.

As mentioned in Section 8.1.6, eight of the subjects watched the original video on the server before rating the seven different cases. Figures 8.10(a) and 8.10(b) show the results categorised based on whether the subjects have seen the preview or not, respectively. The differences between the upgrade and downgrade cases for each bit rate are also not significant anymore. It can also be seen that the subjects who have seen the preview rated the different cases, on the average, lower than the ones who have not seen the previews. However, a pair by pair ANOVA analysis showed that the differences are not significant.

The reason for this trend in lower grades for the preview cases can be attributed to some extent to the fact that the previews were shown on the server




*(a)*Without preview




*(b)*With preview

**Figure 8.10:** *Confidence intervals for the streaming video perceived quality, for subjects who have seen a preview and for those who have not.*

machine, which has a larger screen and better colour depth than that of the tablet computer. Thus, the differences are mainly due to the experimental design (showing the preview video on the server rather than on the tablet PC) rather than user behaviour. This also shows how sensitive the results could be depending upon the experiment design.

The results are also analysed to see the impact of user experience in video streaming on the perceived quality. As can be seen from Figure 8.1(b) in Sec-

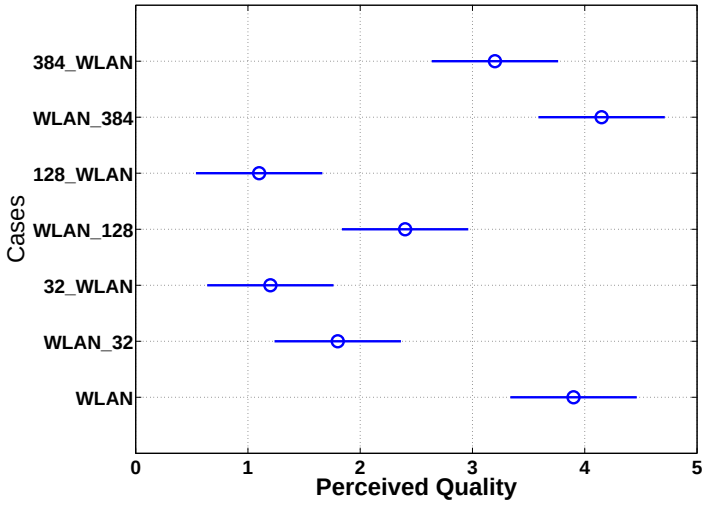



**Figure 8.11:** *Confidence intervals for the streaming video perceived quality for users who frequently use video streaming on the Internet.*

tion 8.1.2, the number of subjects who are used to accessing video streaming via their mobile phones is not large enough to get any statistically significant comparison. Thus, the subjects were categorised based on their streaming experience on the Internet (see Figure 8.1(a)), and Figure 8.11 shows the results for subjects who use streaming frequently. The difference between the *WLAN*_384 and 384_*WLAN* cases, which was mainly attributed to the initial startup buffering time, is not significant anymore. Experienced users of video streaming are aware of the initial buffering requirements, so they are more patient during the startup than non experienced users. However, they still rated the *WLAN*_128 case more than one grade higher than the 128_*WLAN* case because the startup buffering time is more noticeable for the 128 Kbps case (around 30 seconds before the trailer is streamed as compared with the 20 seconds for the 384 Kbps case).

## 8.3 Summary

In this chapter the design and analysis of a usability experiment to find out the user perceived QoS dependency on network characteristics has been in-

vestigated. Experiments have been carried out on the impact of bit rate, FER and RTT on web browsing performance in UMTS and the impact of handover associated bit rate downgrade and upgrade in a heterogeneous network comprised of UMTS and WLAN. The results are gathered through the ratings of forty users and are analysed to spot trends in user quality perception and the relationship between users' ratings and objective quality measures. It is found that it is possible to get meaningful results from the experiment even though only forty users are used. For both the web browsing and streaming services the ratings of the users showed a clear trend and were mostly in line with objective quality measures.

For the web browsing experiments, the effect of the RTT is very perceivable by the subjects, regardless of the web page size. On the other hand, the effect of FER is found to be significant only for larger pages.

The streaming experiments showed that the initial startup buffering time of the video is very important for the users, no matter what the quality of the streaming is once it has started. Users are also very sensitive to any re-buffering that occurs after the streaming has started, and they rated the overall quality regardless of the video quality after the re-buffering interruption. Users who are experienced in video streaming are found to be less critical of the start up time than users who are not used to the service.

# Part VI

# Conclusions

# Chapter 9

# Conclusions and Discussions

The overall goal of this PhD study has been the investigation of packet service performance in UMTS networks, both in standalone and heterogeneous settings. We have looked into the problem from several aspects to get a holistic view. Specifically, the impact of transport and link layer protocols, vertical handover and perceived service quality has been the main focus of the investigation. The performance results have been generated using analytical modelling, emulations and usability experiments.

In the following sections, a summary of the whole thesis is given, drawing the main findings from the most relevant topics investigated. Finally, some pointers regarding future work are given.

## 9.1 Performance Impact of RLC Reliability Mechanisms on TCP

The impact of the different parameters that control the UMTS RLC protocol's reliability has been investigated using analytical modelling and emulations. It has been shown that by using only a subset of these parameters a deadlock free link layer communication can be set up.

The maximum number of RLC retransmissions is the main factor that determines the performance. When the operating FER is 10%, setting the maximum number of retransmissions to three or more eliminates TCP timeouts and gives optimal performance, decreasing the file download time by around 85% as compared with the case with no link layer retransmission. RLC's retransmission mechanism is also very efficient, with an overhead of less than 5% for status reporting.

Though allowing higher number of RLC retransmissions leads to higher RTT, it is no disadvantage for the cases investigated here as the main performance metric of FTP is the total file download time. Also TCP timeouts are prevented. Even for services such as web browsing where the RTT matters more, the increase in RTT due to many RLC retransmissions is not a problem as long as TCP timeouts do not occur. However, allowing higher number of RLC retransmissions can be a disadvantage if the FER is high. This is because TCP can timeout before the RLC manages to recover the errors. In this case, valuable bandwidth is wasted due to redundant RLC retransmissions.

## 9.2 Performance Impact of TCP Spurious Retransmissions

The impact of spurious TCP retransmissions caused by packet reordering and delay spikes and the performance enhancements due to the Eifel and DSACK TCP enhancements has been investigated.

Eifel is very effective in mitigating spurious retransmissions due to packet reordering, increasing the goodput by up to 33%. DSACK also shows performance improvement against packet reordering but the goodput improvement is not more than 14%. The usage of SACK (without DSACK) causes performance degradation because there is a higher probability of entering the TCP fast retransmission phase earlier when using SACK as the sender does not have to wait for triple duplicate ACKs. Eifel also improves performance in the case of spurious retransmissions caused by delay spikes but the improvement

in this case is not more than 9%. The reason for this is that higher delay values trigger multiple timeouts, in which case the Eifel algorithm acts conservatively in the same way as a normal slow start.

The Eifel algorithm is currently being standardised by the IETF [81; 125] and the TCP timestamp option, which is the recommended way to enable the Eifel algorithm, is already widely deployed in the Internet [95]. Eifel is also part of the Linux TCP, which is the most widely used operating system for web servers [96]. On top of this, Eifel is a sender side algorithm, the only requirement on the receiver being enabling the TCP timestamp option. Considering these factors, Eifel is a very practical solution to cope with TCP spurious retransmissions due to packet reordering. Thus combining RLC out of sequence delivery with Eifel is a suitable setting for services such as web browsing where RTT is an important performance metric. With regard to Eifel's relatively poorer performance against delay spikes, an enhancement where slow start phase is not entered even after multiple timeouts is already proposed in [90].

## 9.3 Performance Impact of Vertical Handover

UMTS networks are currently deployed in environments where a GPRS network already exists and WLAN is becoming very accessible via WLAN hot spots. Thus, the performance of FTP file downloads in a heterogeneous network comprised of UMTS, WLAN and GPRS has been investigated. The behaviour of ordinary TCP (new-Reno), Eifel, and SACK under different handover conditions is studied and a new buffering scheme called intermediate buffering, where only packets that arrive during handover are buffered, is proposed. This buffering scheme is compared with full buffering, where all the packets in-flight when the handover is started are buffered in addition to the packets that arrive during the handover, and also with no buffering schemes.

For the no handover delay cases, full buffering performs the best. For non zero handover delay cases and ordinary TCP, intermediate buffering is the best

performer (goodput improvement in the range of 15%), except in the case of a handover from GPRS to UMTS (goodput decreased by up to 12% for this case). Full buffering performs even worse than no buffering for the upward handover (handover from a small coverage to a larger coverage network, e.g. UMTS to GPRS) cases, decreasing the goodput by up to 15%. However, it performs better than no buffering for upward handover cases, with up to 13% improvement in the goodput.

Eifel degrades the performance of intermediate and no buffering, decreasing the goodput by up to 34% for the UMTS to WLAN handover case. However, Eifel improves the performance of full buffering, with up to 12% improvement in the goodput.

SACK has no effect on full buffering. If there is a handover delay, SACK also has no effect on no buffering. However, when there is no handover delay, SACK has a positive impact (up to 7% improvement in goodput) for the downward handover cases and negative impact (up to 30% decrease in goodput) for the upward handover cases. For intermediate buffering, SACK has a positive impact on the performance in the downward handover from GPRS to UMTS, increasing the goodput by up to 11%.

From the results it can be seen that there is no buffering mechanism or TCP extension that is optimal for all the considered cases. However, we recommend the use of intermediate buffering as it gives the best result in the most realistic cases, i.e. all except the no handover delay case. It also places less memory requirements on the access points as compared with the full buffering scheme.

## 9.4 User Perceived Quality of Packet Services

Quality of service in mobile telecommunication systems is usually identified by some basic performance metrics such as delay, throughput and jitter. However, the main impact of service quality is on the end user, and thus we found it important to undertake a detailed study of service performance using usability experiments. Experiments have been carried out on the impact of bit rate,

FER and RTT on web browsing performance in UMTS and the impact of handover associated bit rate downgrade and upgrade in a heterogeneous network comprised of UMTS and WLAN.

Meaningful results have been found from the experiments even though only forty test subjects were used and the subjective results are in line with the objective measures of quality. For the web browsing experiments, the effect of the RTT is very perceivable by the subjects, regardless of the size of the web page. On the other hand, the effect of FER is significant only for larger pages. The streaming experiments showed that the initial start up buffering time of the video is very important, no matter what the quality of the streaming is once it has started.

Results that are based on objective performance quality metrics can be augmented with subjective evaluations to decide on whether to use a certain network/protocol optimisation or not. For example, by looking at the results from the packet reordering studies (Figure 6.7) and the web browsing quality vs. RTT results (Figure 8.7), the recommendation to use Eifel with RLC out of sequence delivery is reinforced.

## 9.5 Future Work

This thesis has provided the evaluation of packet service performance in a standalone UMTS network and a UMTS network coexisting with WLAN and GPRS. It would be quite interesting to extend the performance investigations for UMTS evolutions such as HSDPA whose technology has become mature enough to be deployed. This requires the enhancement of the emulation platform used in this PhD study to support HSDPA functionalities such as Hybrid ARQ (H-ARQ) and link adaptation.

TCP was designed for wired networks where congestion is the major performance bottleneck. Though there have been tens (if not hundreds) of extensions to optimise TCP performance in wireless networks, apart from the ones discussed in this thesis, there seems to be no consensus in the network-

ing community regarding which ones to adopt. Not only that, what one proposal improves might be already undone by another proposal, because there is no standardised way of joint comparison. Thus, it would be of utmost importance to have a standardised way of testing new protocol enhancements. Such a platform should consider the different aspects of the network, through which researchers can test if their new protocol enhancements are performing as expected in different scenarios and if they conflict with other existing enhancements.

In the context of vertical handover, current research is mostly concerned with network level optimisations, e.g. handing over to the network with the best signal strength. However, these network level and system level enhancements may not directly translate to user satisfaction. Thus, it would be interesting to investigate utility functions that also consider the user (in terms of several factors such as cost, mobility and urgency of the task being performed) and the service at hand (service delay and bandwidth requirements, for example). An enhanced system level usability testbed could be very useful for such investigations. We envision a usability test setup which provides the test subjects with a dynamic interface, instead of the fixed network setup used in the usability experiments conducted in this PhD study, where they can change the network behaviour such as the bit rate, depending on their preference in terms of cost and quality. Multiple test users can be assessed concurrently and an invaluable information regarding users' cost-quality trade off can be gathered, which then can be used for defining optimal utility functions for handover decisions.

Throughout the study, we assumed a congestion free environment, as the focus was solely on the wireless network. However, if the application servers reside in the wired Internet domain, congestion is imminent and future investigations should also consider the impact of congestion in addition to the impact of the wireless network.

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# Part VII

# Appendix

# Appendix A

# Basic Networking Concepts

A data communication network is an interlinked system of devices, also known as *nodes*, that is used to transfer data from a sender to a receiver(s). Data communication can be realised in either a Circuit Switched (CS) or Packet Switched (PS) fashion, or their combination [126; 127].

In CS, a path from source to destination is selected when the data session is started, and it is kept throughout the session. However, the links can be shared by several connection with a multiplexing scheme. Traditional fixed telephone networks employ CS. On the other hand, in PS, the data is divided into discrete, potentially of varying length, blocks called *packets* and the route is dynamically selected, possibly for each packet. Each packet contains a *header*, which is a control information regarding source/destination addresses, and possibly additional information such as priority. The packets are routed at each node in the End to End (E2E) path based on the header information. The Internet mainly employs PS[1].

Due to their complex nature (the different types of media such as wired and wireless, different application types etc.), networks are usually designed in a layered architecture that divides functionality into manageable units called

[1] PS was originally suggested for the Internet due to its bandwidth efficiency and robustness. However, recent studies have shown that PS is inefficient in high speed Internet backbones, and CS alternatives are being proposed [128].

*protocol layers*. The most widely used layered protocol architecture is the Open Systems Interconnection (OSI) reference model [44].

Figure A.1 illustrates the OSI model. Each layer utilises the services of the layer just below it, according to some defined *interface*. A transmitting protocol entity receives data units, known as Service Data Units (SDUs) from the layer above it. The SDUs are then processed and passed on to the underlying layer in a modified format (the most common changes being segmentation and the addition of header information) called Protocol Data Units (PDUs). The transmitted PDUs are identical to the SDUs expected at the underlying layer and thus a PDU transmitted from layer $N$ become an SDU in layer $N-1$. The receiving entity converts one or more PDUs into an SDU and passes the SDU to the layer just above it.

The bottom three layers (physical, data link and network, also known as layer 1, layer 2 and layer 3) are concerned with network services, i.e. moving data from one device to another [126; 127]. For this reason, they are commonly referred to us *lower layers*. The physical layer is responsible for

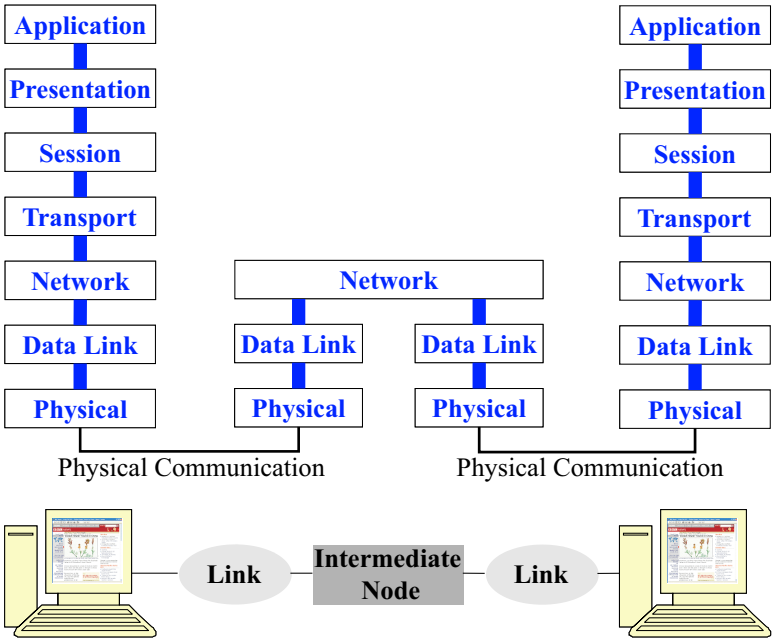


**Figure A.1:** *The OSI reference model (adapted from [127]).*

transmitting bits over the physical medium. The data link layer, by utilising the services provided by the physical layer, coordinates the transfer of frames of data between directly connected nodes. Most link layers also provide error detection and correction schemes. The network layer is responsible for routing packets from source to destination, often through multiple links, and possibly across several networks.

The upper four layers (transport, session, presentation and application, also known as layers 4-7) are concerned with providing services to the end user and thus not concerned with the network[2] [126]. They are commonly referred to as *higher layers*. The transport layer is concerned with E2E delivery of entire messages (as compared with the network layer, which treats each packets independently). The session, presentation and application layers are usually lumped into one in many implementations (such as in the TCP/IP protocol stack implementation that the Internet is based on [49]) and they provide the functionalities of service session management and application interface through which a user can access the services provided through the network.

The performance of networks, the links connecting nodes within the network and also those that interconnect several networks are usually specified with some key performance parameters. The most important ones are the *data rate* and *delay*. Other parameters such as number of hops, load, reliability, and Maximum Transmission Unit (MTU) (which is the maximum packet size that can traverse a link/network) are also used to characterise a link/network.

The data rate, also referred to as *bandwidth* or *bit rate*, is the maximum amount of bits that can be put over a link in a given time[3]. The delay (also known as *latency*), on the other hand, specifies how long it takes for a certain packet to traverse from the source to the destination and it is composed of propagation delay (which is negligible in most terrestrial networks), queueing delay, processing delay (such as framing, compression and error detection),

[2]This classification is a little bit blurred, specially when it comes to the transport layer, because some transport layers are optimised for a specific networks.

[3]Note that a given interface to a link can support multiple data rates and a given data rate is selected depending on several factors such as the load in the system and the user's Service Level Agreement (SLA) with the network provider.

and transmission time. The transmission time depends on the data rate and packet size and it is simply the packet size divided by the data rate.

Several other parameters, most of which derived from the data rate and delay, are also commonly used to characterise network performance. The most notable ones are:

- **RTT:** This refers to the time it takes for a packet to travel from the sender to the receiver and back, not necessarily using the same path. If the path and the packet size used in both direction are the same, then the RTT is simply twice the latency.
- **Packet loss ratio:** This refers to the percentage of packets that are received erroneously or not received at all. Several factors such as corruption in the transmission media (specially wireless links) and congestion cause packet loss.
- **Throughput:** This refers to the total number of bits transferred over the network in a given time. Note that the difference with the data rate is that throughput is the actual rate of transfer, not the maximum possible. Apart from the data rate and amount of data available to be sent, the throughput also depends on the behaviour of protocols, such as transport layer protocols that provide congestion control mechanisms.
- **Goodput:** This is similar to the throughput, but it refers only to actual transfer rate of user data (i.e. excludes overheads such as packet headers, signalling packets and retransmissions).

# Appendix B

# Real Time Emulation Platform

As mentioned throughout the report, the results in this thesis are generated using Real-time Emulator for Service Performance Evalution of Cellular neTworks (RESPECT). The following sections describe the design and implementation of RESPECT, and its extended version that also supports heterogenous network emulations. For a detailed description of RESPECT, the reader is referred to [72].

## B.1 Core components

RESPECT is composed of four core modules, each module containing a set of interrelated classes. The four different modules are called *Common, Framework, Sniffer* and *Emulator*. The Common, Framework and Sniffer modules are generic modules that can be used to extend the emulator to represent other networks. The Emulator module is a collection of classes, mostly inherited from the Framework and Common modules, that specifically model the UMTS protocols.

### B.1.1 Common Module

The common module provides base classes that are used to define interfaces for the different parts of the system. This module contains five main classes, namely *Data_Handler*, *Data_Source_Sink*, *Thread*, *Configuration_Reader* and *Logger*. Data_Source_Sink provides an interface for entities that act as sources or sinks of data. The Data_Handler class, on the other hand, provides an interface for those entities that are responsible for handling data. The Data_Handler and Data_Source_Sink are associated 1-to-1, i.e. every source/sink should have a handler that handles its data, and every handler should have an associated source/sink where it gets its data and where it passes the data after it has processed it. The Configuration_Reader is a simple parser that handles parameter files. Thread is an abstract class that provides a wrapper for creating and configuring the priority levels and the scheduling policies of threads using C++ classes. The Logger class is used for logging data, such as results from emulation runs and debugging information about the emulator operation.

If extensive logging to the hard disk is to be done at the same time as the emulation process, timing requirements might not be met, as writing to the hard disk is very slow and takes a considerable amount of CPU time compared to emulation event periods. The logger mitigates this drawback of logging using remote logging (all logging is done via a TCP connection on a remote server, i.e. no local hard disk access), accompanied by priority scheduling (where the logging activities are set to the lowest priority) and voluntary processor yielding.

Figure B.1 shows the impact of remote logging and prioritisation. When the logging is done directly to the hard disk (Figure B.1(a)), deadlines can be missed by hundreds of milliseconds, even though the mean time difference is very small. Such rare but high delays can have a big impact for high bit rate connections. By performing the logging remotely, the maximum timing mismatch was reduced to a large extent, as shown in Figure B.1(b). The mean time difference is higher than the previous case, but as the mean times in both cases are very small, the outliers are the ones that affect the system performance the most. When remote logging is combined with the aforementioned

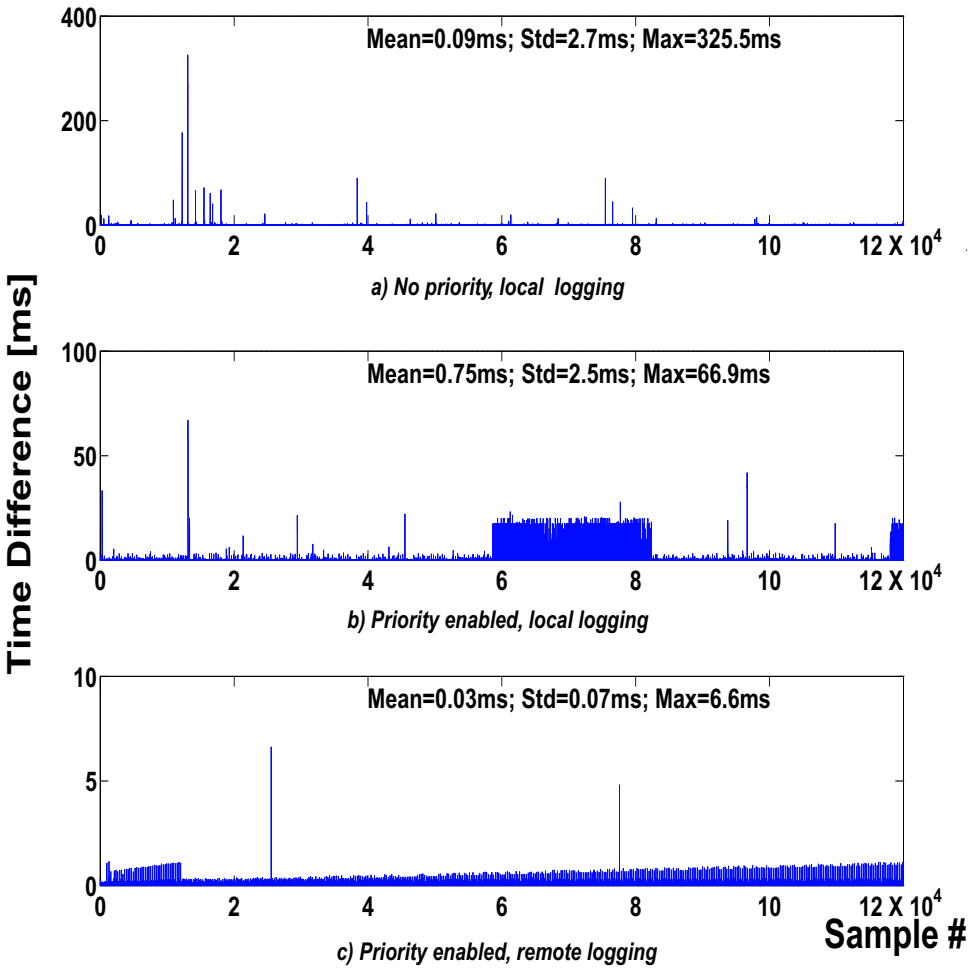


**Figure B.1:** *The impact of prioritisation and remote logging on meeting time expiration deadlines. The Y axis shows the difference between the actual time an action was taken and the scheduled time.*

priority scheduling scheme, the timing mismatches, in terms of the mean, standard deviation and the maximum values are improved tremendously, as shown in Figure B.1(c).

## B.1.2 Sniffer Module

The *Sniffer* module provides the functionality of intercepting incoming/outgoing packets from/into the network on a real-time basis and supplies the data

to the emulator module, which handles the packets and gives a decision on what is to be done. The interception and re-injection of packets is done using the Netfilter/Iptables packet handling framework [129]. Using this framework, it is possible to drop, accept or queue packets based on criteria such as IP addresses, port number or packet sizes.

During its setup, the Sniffer initialises the Netfilter packet filtering rules for queueing the concerned packets. When a packet arrives at the IP layer in the emulator, the Sniffer raises an event to its associated data handler, which is the emulator, letting it know that new data has been received. This event includes information such as the size, identification and the direction of flow of the packet, i.e. whether it is incoming or outgoing. The packet will be kept inside the queue till the Sniffer receives a verdict to discard or release the packet from the emulator. The Sniffer also logs the header information of every incoming/outgoing packet in a *pcap* file format so that the connection could be analysed further by using powerful tools such as *tcptrace* [130].

### B.1.3 Framework Module

The core protocol components of the emulator are designed based on CASINO (Component Architecture for Simulating Network Objects) [131]. CASINO is an object-oriented protocol development platform that is targeted for developing network communication protocol stacks. A scaled-down version of the CASINO framework, with some modifications to fit the needs of RESPECT, was re-implemented, based mainly on the design specifications of CASINO [132]. The *Framework* module provides generic classes that implement basic functionalities of protocols, packets, events, and timer handler that control the events.

### B.1.4 Emulator Module

This is the module that actually performs the UMTS emulation process. The main class in the emulator module, known as *Emulator*, initialises the different

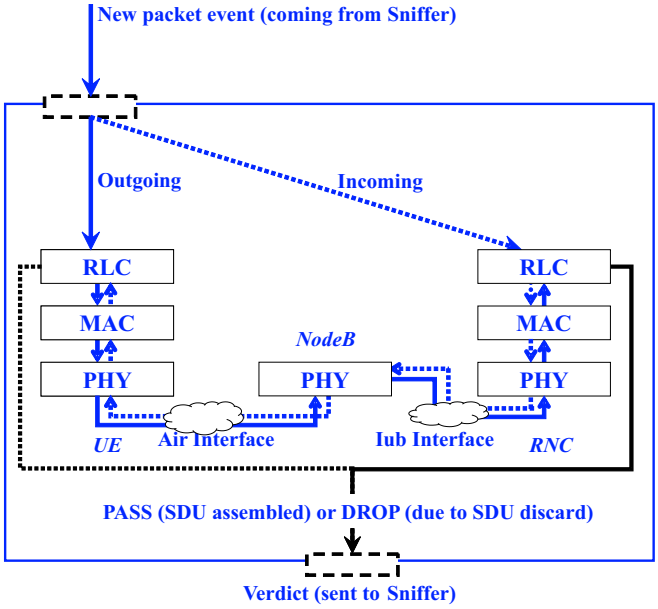


**Figure B.2:** *Communication between Emulator and Sniffer.*

protocols and network entities and interconnects them for proper packet flow. To get the emulation process started, the initial timers that would keep on firing every Transmission Time Interval (TTI) are started. The Emulator accepts packet from the Sniffer, propagates them down the protocol stack and also notifies the Sniffer on behalf of its components when packets are dealt with, i.e. when it has passed through the whole protocol stack and ready to be released back to the network or when a drop decision is made due to errors in the air interface or buffer overflows. Figure B.2 shows this communication process between the Sniffer and Emulator.

Apart from the Emulator class, the other main classes in this module are the RLC, Medium Access Control (MAC), Physical (PHY) and Air Interface. The RLC is implemented following the 3GPP RLC specifications [61], with full support of the retransmission facilities described in the specification. Figure B.3 summarises the RLC procedures and the details of each sub task is given in detail in Figures B.4 - B.6.

The MAC layer is the one that determines the rate of information flow and the granularity of the emulation process. Every TTI, the MAC sends informa-

tion to the RLC, telling it the amount of data that could be accommodated in one TTI, the length of subsequent TTIs and the PDU sizes that the RLC should use next time it segments packets.

The PHY layer is a simple class that passes packets coming from the MAC to the air interface and vice versa. The air interface provides a memory-less channel that has some given FER (frame erasure rate) in the UL and DL directions and associated processing delays in both directions.

## B.2 Support for Heterogeneous Networks

For the studies conducted in Chapters 7 and 8, RESPECT was modified to accommodate the emulation of heterogeneous networks. Figure B.7 shows the structure of this new extension. The detailed UMTS protocol stack implementation that was shown in Figure 5.9 is removed, and instead the network behaviour is described as a state machine.

The state machine can be defined to operate either probabilistically or deterministically. A two layered Markov Chain (transition between different networks and sub-state transitions within a given network) are used to specify the probabilistic transitions. There are two timers that drive the emulation process: the network transition timer and the state transition timer. When the network transition timer fires, whose granularity is in the order of few seconds

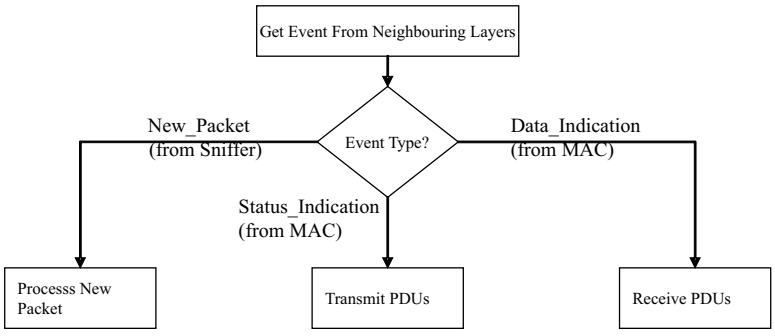


**Figure B.3:** *RLC processing summary.*

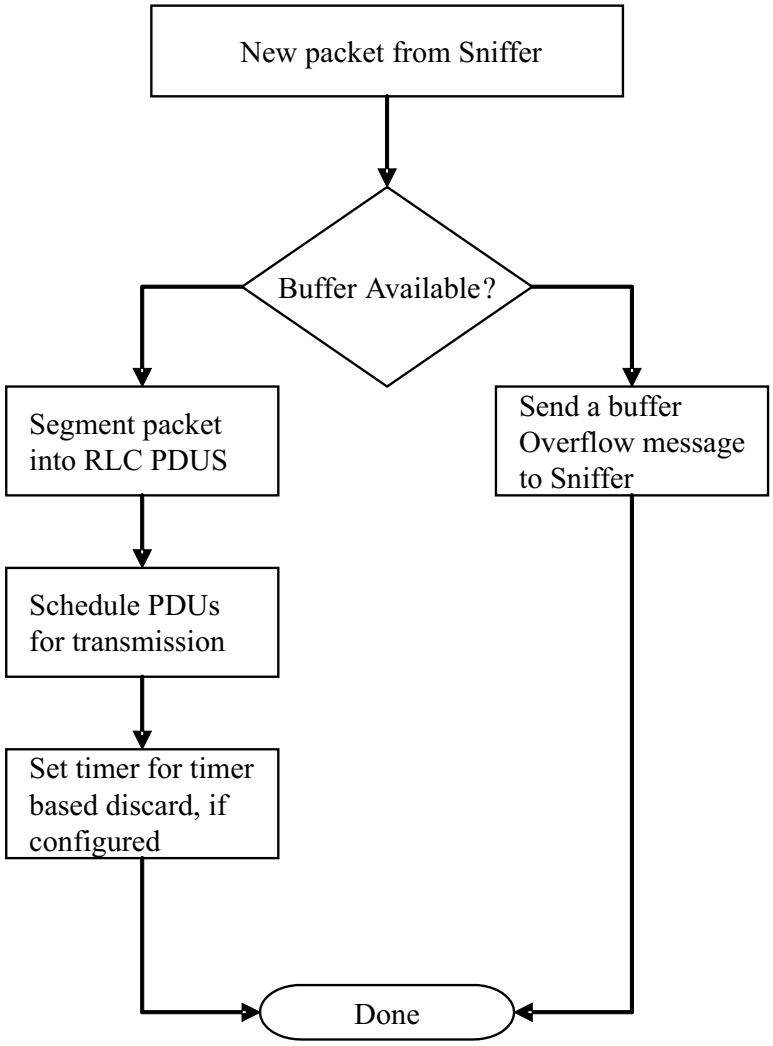


**Figure B.4:** *Arrival of new packets.*

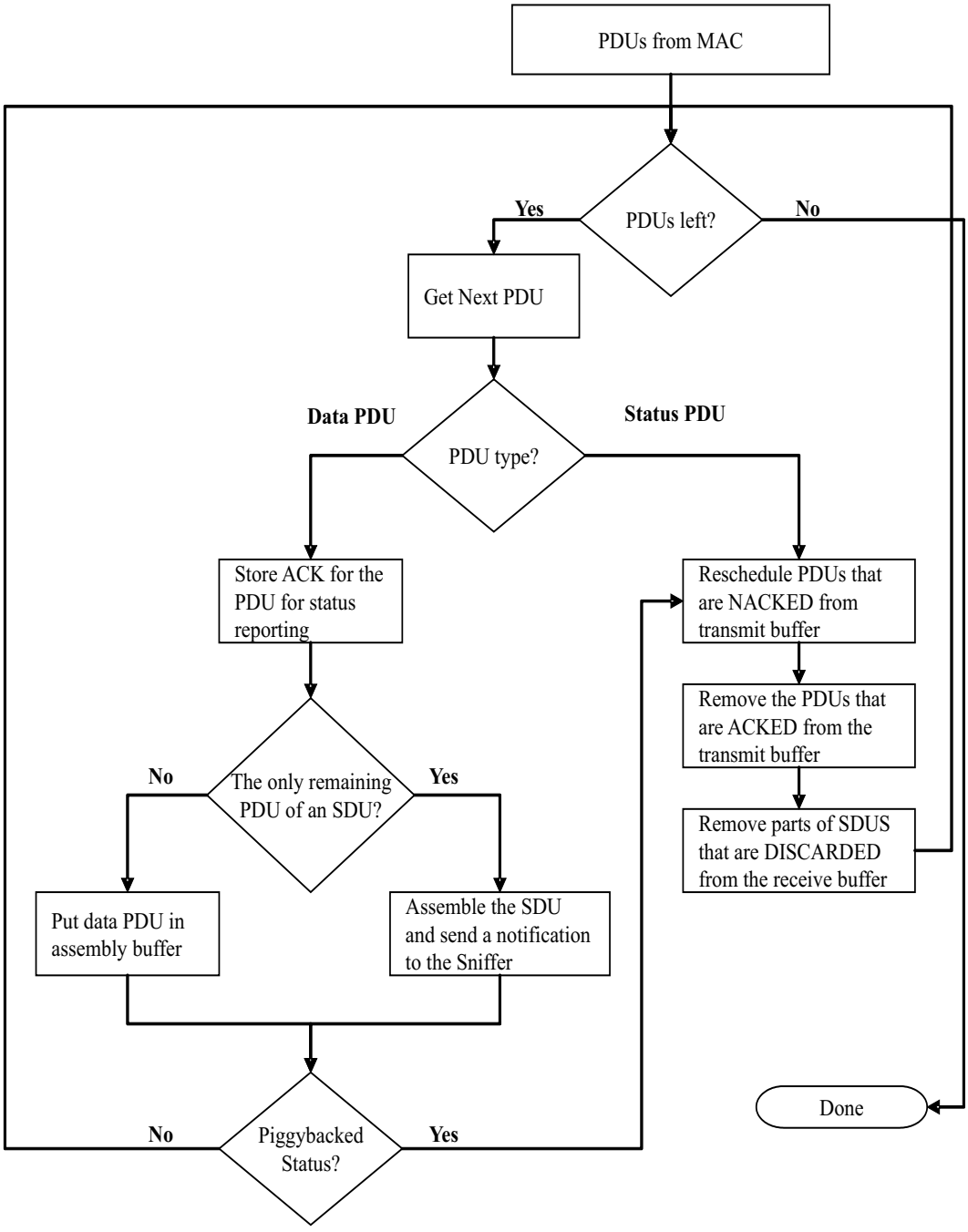


**Figure B.5:** *Reception of PDUs.*

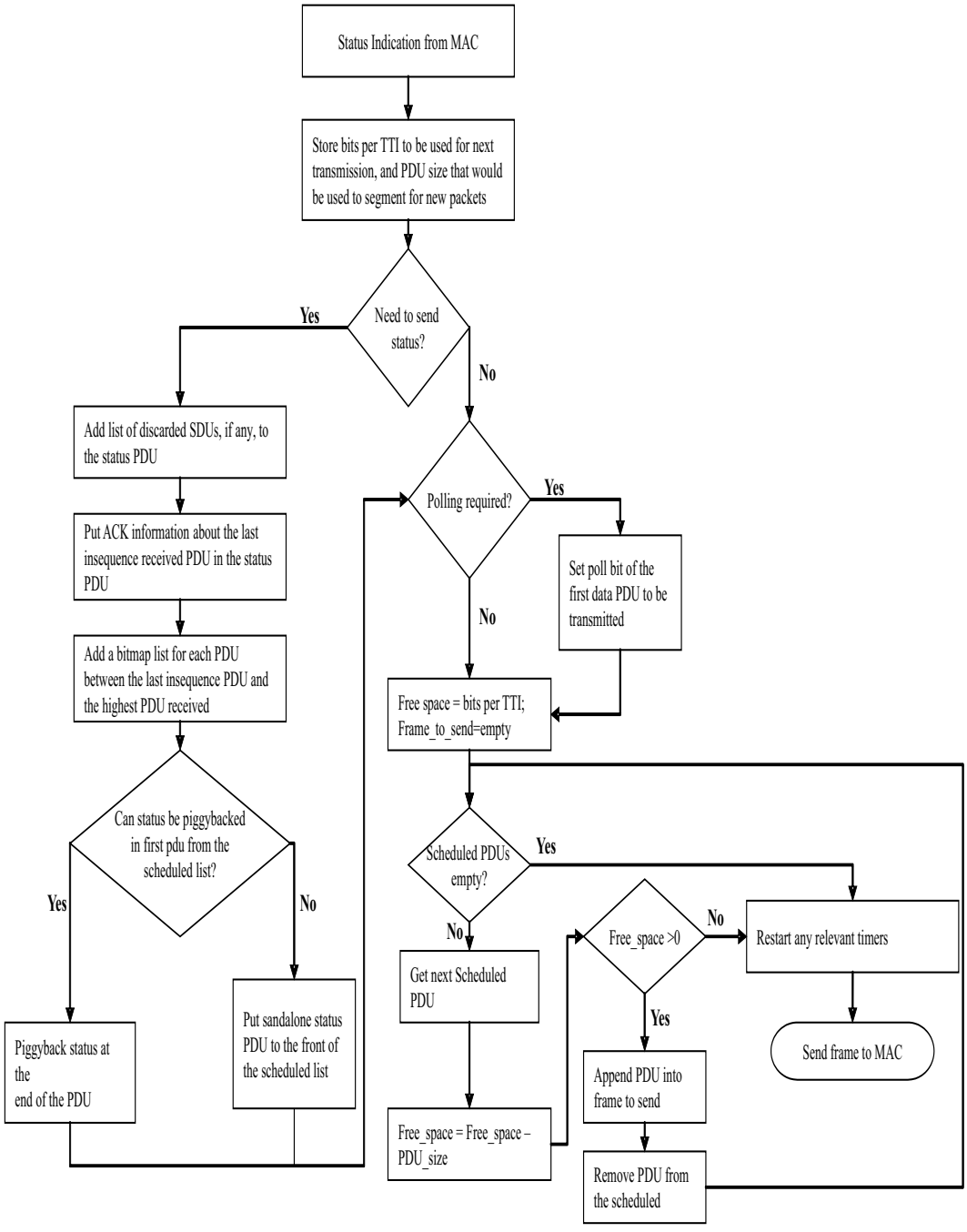


**Figure B.6:** *Transmission of PDUs.*

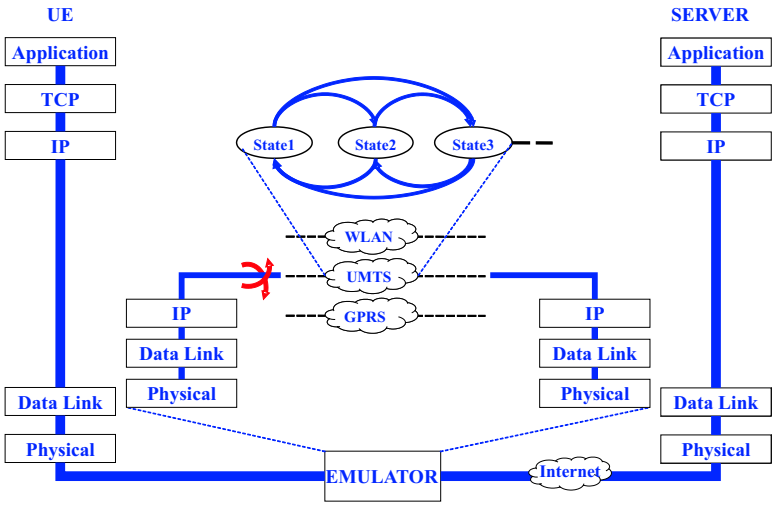


**Figure B.7:** *RESPECT extended for heterogeneous networks.*

(which depends on the user's mobility, fast movement resulting in lower timer values), decisions will be taken whether to remain on the same network, or handover to another network, based on the network transition probabilities. On the other hand, the state transition timer, with finer granularity than that of the network transition timer, specifies when to check for transitions between the states of a given network (basically bit rate downgrades and upgrades).

A time limit or amount of data transmitted can also be used to specify when a transition should be made between networks or within the sub-states of a given network. Since the aim of this project is on link level studies and as recent studies have found that Markov chains have low accuracy in capturing network transition and sub-state transitions [133], the deterministic way of specifying network transitions was used for the investigations carried out in this thesis.

As shown in Figure B.7, the emulator intercepts packets at the IP level. Based on the current network condition (current bit rate, delay distribution, other temporary conditions such as handover and outage), packets will be marked with a timestamp and they will be queued. The timestamp specifies at what time the packets should be released to their destination. When handover takes place between different networks, there is a possibility of just pausing the connection (not the source traffic flow, as the emulator is transparent to the application) or flushing the queued IP packets.

## B.3 Accuracy of Results

Network simulations are commonly divided into two categories *finite time horizon* (also called terminating) and *steady state* [134]. In finite time horizon the interest is in the behaviour of the network within a well specified period. On the other hand, in steady state simulations the stable behaviour of the network is the main interest. While it can be quite complicated to verify the accuracy of steady state simulations, a terminating simulation's accuracy can easily be guaranteed by repeating the simulation with different seeds and stop

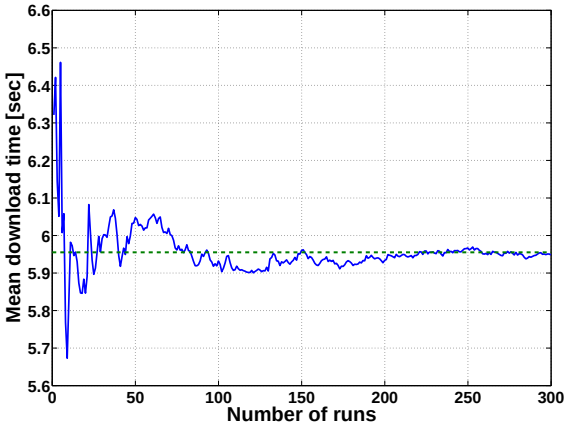


**Figure B.8:** *Evolution of the mean download time with the number of runs.*

when the accuracy is enough for the task at hand [134].

RESPECT is intended to be used in finite time horizon cases where the interest is the performance of the link during a given duration that it takes to perform a certain task (for example, file download). And as such, for the experiments carried out in this thesis, the file downloads were repeated several times, each with a different random seed, until the mean value of the performance metric considered stabilises.

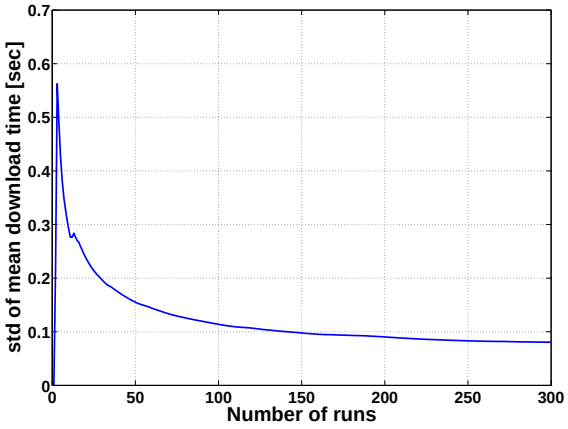


**Figure B.9:** *Evolution of the standard deviation of the mean download time with the number of runs.*

Figures B.8 and B.9 show the evolution of the mean download time and its standard deviation for one of the cases investigated in Section 6.4.1 (the No Eifel, out of order delivery sample in Figure 6.3). As can be seen from the figures, the mean stabilises for a hundred or more number of simulations, and its standard deviation decreases very fast with the number of samples, becoming less than 0.1 seconds when the number of runs is more than 150.

Figure B.10 shows the box plot of the normalised mean download time for different number of simulation runs. As can be seen from the figure, even with 10 simulations, for 50% of the cases (i.e. the ones in the inter-quantile range) the error is less than 2%. The errors become marginal as the number of simulation runs is increased, reaching below 0.5% for 300 simulations. The result is similar for the other cases investigated in Chapters 5 and 6.

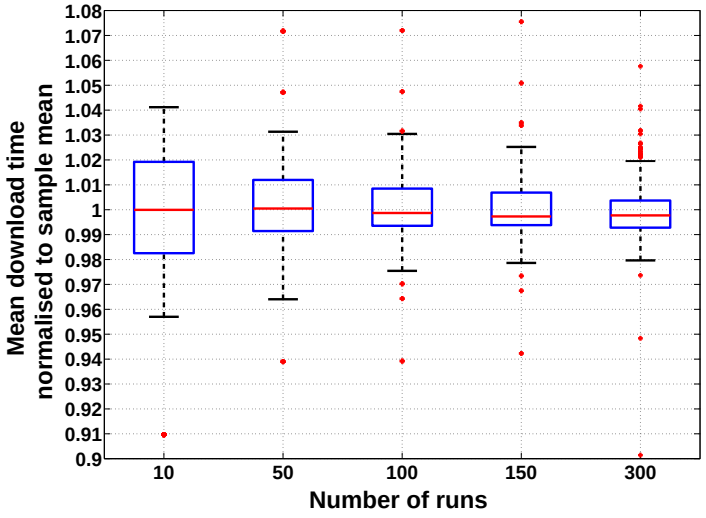


**Figure B.10:** *Box plot of the mean download time normalised over the sample mean.*

# Appendix C

# Setting RLC Polling Parameters, an Example

In this appendix, an example illustrating a deadlock occurring in the RLC, and how it can be corrected is given.

A 100 Kbyte file was being downloaded several times, for a 384 Kbps connection with an average FER of 10%, to get some statistical averages on the file download times. *Poll last PDU*, *Poll last retransmitted PDU*, and *missing PDU indication* are all enabled to control polling and status reporting. It was noticed that some of the values were quite large (in the order of ten times the mean). Tables C.1 and C.2 show the traces from the RLC and TCP layers at the UE side, for one such case (the negative ID numbers refer to status PDUs).

From Table C.1 it can be seen that PDUs #1 & #2 of SDU# 15 were erroneous when they were received at TTI #488. So the polling request by the sender, as identified by the polling bit of PDU #1 is lost along with it. Nothing was being received between TTIs #497 and #787, that is for 2.9 seconds, as each TTI represents 10 ms. This is confirmed by the TCP trace where there is also no activity for almost 3 seconds (time difference between rows #2 and #3 in table C.2).

This idle period starts after the transmission of the SYN packet (which

**Table C.1:** *Traces from RLC layer.*

| TTI | (SDU#, PDU#) | #Transmission | Polled | Erroneous |
|---|---|---|---|---|
| 469 | (14, 2) | 1 | 0 | 0 |
| 487 | (-1, 15) | 1 | 0 | 0 |
| **488** | **(15, 0)** | **1** | **1** | **1** |
| **488** | **(15, 1)** | **1** | **0** | **1** |
| **786** | **(-1, 16)** | **1** | **0** | **1** |
| 787 | (16, 0) | 1 | 1 | 0 |
| 787 | (16, 1) | 1 | 0 | 0 |
| **802** | **(15, 0)** | **2** | **1** | **1** |
| **802** | **(15, 1)** | **2** | **0** | **1** |
| 887 | (17, 0) | 1 | 1 | 0 |
| 887 | (17, 1) | 1 | 0 | 0 |
| 901 | (15, 0) | 3 | 1 | 0 |
| 901 | (15, 1) | 3 | 0 | 0 |
| 918 | (-1, 17) | 1 | 0 | 0 |

corresponds with SDU# 15 at the RLC layer) of the TCP connection setup procedure. The deadlock occurred because the SYN packet was the last packet in the window on the sender's side and the sender TCP will not be transmitting any data packets till it is done with the connection setup process, i.e. an ACK for the SYN is received from the other side. The loss would not have caused a major drawback if the packet that was lost laid somewhere in the middle

**Table C.2:** *Traces from TCP layer.*

| | |
|---|---|
| 1 | 14:57:10.209868 UE > SERVER: S 1736209341:1736209341(0) |
| 2 | 14:57:10.210397 SERVER > UE: S 2037954080:2037954080(0) ack 1736209342 |
| 3 | 14:57:13.202027 UE > SERVER: S 1736209341:1736209341(0) |
| 4 | 14:57:13.202560 SERVER > UE: S 2037954080:2037954080(0) ack 1736209342 |
| 5 | 14:57:14.212273 SERVER > UE. S 2037954080:2037954080(0) ack 1736209342 |
| 6 | 14:57:14.523085 UE > SERVER: . ack 1 |
| 7 | 14:57:14.544901 SERVER > UE: . 1:1449(1448) ack 1 |

of the transmission window, as missing PDU indication would have triggered status reports when a higher sequence numbered PDU is received. Finally, TCP times out and there is a TCP retransmission. The retransmitted SYN corresponds to SDU# 16.

Apart from this deadlock, it can also be seen that the RLC keeps trying to send SDU# 15 even though TCP has given up waiting for an ACK and retransmitted the packet. When the receiver gets the PDUs corresponding to the retransmitted SYN packet at TTI #787, the missing PDU indication triggers the transmission of a status report. The sender responds by retransmitting the PDUs that were lost, which are unfortunately received erroneously once again at TTI #802. This shows that the deadlock not only slowed the connection by preventing the sender from transmitting for 3 seconds, but also led to the simultaneous retransmission of a packet at the TCP and RLC layers, and hence wasting the precious wireless bandwidth.

Using a timer based polling resolved this deadlock that happens when the last PDU in the transmission buffers is lost. This is because the sender starts a timer when the last PDU is sent as a polling bit is set along with it due to the poll last transmitted PDU setting. If no ACK or NACK regarding this PDU is not received when this timer expires, the sender retransmits the last PDU in order to resend the lost poll. Hence there is no need to wait for a TCP timeout to continue data transmission.

# Appendix D

# Usability Experiment Forms

In this appendix, the orientation script and questionnaires used in the usability experiments are given.

## D.1 Orientation Script

This orientation script was read out at the beginning of each experiment session to ensure that all users get the same information regarding the experiment.

"Hi, I am Oumer Teyeb. Welcome to this usability testing session and thanks for taking the time.

We will be testing the effect of a mobile network's characteristics on the perceived quality of services of web browsing and video streaming services. You will be accessing the services using a tablet PC, but you have to assume that you are accessing the services over your mobile device.

In order to make the results of your observations as unbiased as possible, we will be conducting blind tests, i.e. you will not be told the different network settings that will be tested.

While performing the tests, you will also be filling a questionnaire. After

each task, you will give a score regarding the quality of the service you just experienced. There is also a space for putting additional comments for each task.

I will be sitting next to you and we will conduct the tests together. I will change the network settings, you perform the corresponding task, grade the quality you just experienced, and then we go to the next item to test and so on.

Do you have any questions?"

## D.2 Questionnaire

A snippet of the questionnaire that the users filled while performing the different tasks during the usability experiment is given below.

**TASK 1:** When the test conductor asks you to do so, click on the shortcut in the desktop named “Google". If you are not sure how the download went, press “ctrl+F5" to force a refresh. Notice how the overall web page is loaded. After the page has loaded you can press the “search" button to go to the next page. Once this is done, close the Internet Explorer window, grade the quality of the download you just experienced and wait for the test conductor’s instruction to repeat the same procedure. A total of eight repetitions are made.

| Run # | Experienced Quality | Any additional comments |
|---|---|---|
| 1 | A. Excellent<br>B. Good<br>C. Acceptable<br>D. Poor<br>E. Unacceptable | |
| ..<br>..<br>.. | ..<br>..<br>.. | ..<br>..<br>.. |
| 8 | A. Excellent<br>B. Good<br>C. Acceptable<br>D. Poor<br>E. Unacceptable | |

**TASK 2:** When the test conductor asks you to do so, click on the shortcut in the desktop named “Nokia". If you are not sure how the download went, press “ctrl+F5" to force a refresh. Notice how the overall web page is loaded. Once the page is loaded properly (or if you have already made your decision about the quality before the page is completely loaded) close the Internet Explorer window, grade the quality of the download you just experienced and wait for the test conductor’s instruction to repeat the same procedure. A total of eight repetitions are made.

| Run # | Experienced Quality | Any additional comments |
|---|---|---|
| 1 | A. Excellent<br>B. Good<br>C. Acceptable<br>D. Poor<br>E. Unacceptable | |
| ..<br>..<br>.. | ..<br>..<br>.. | ..<br>..<br>.. |
| 8 | A. Excellent<br>B. Good<br>C. Acceptable<br>D. Poor<br>E. Unacceptable | |

**TASK 3:** When the test conductor asks you to do so, click on the shortcut in the desktop named “Movie_Trailer", which will start a real player streaming video. Once the video clip is finished close the real player and Internet explorer windows and wait for the test conductor’s instruction to repeat the same procedure. A total of seven repetitions are made.

| Run # | Experienced Quality | Any additional comments |
|---|---|---|
| 1 | A. Excellent<br>B. Good<br>C. Acceptable<br>D. Poor<br>E. Unacceptable | |
| ..<br>..<br>.. | ..<br>..<br>.. | ..<br>..<br>.. |
| 7 | A. Excellent<br>B. Good<br>C. Acceptable<br>D. Poor<br>E. Unacceptable | |

# Appendix E

# Example Illustrating Eifel Performance Degradation

This appendix describes in detail the behaviour of the Eifel algorithm as shown in Figure 7.7 of Section 7.3.1.2. The figure, which shows the time sequence graph where Eifel is used with no buffering and no handover delay, is shown here again for convenience.

As can be seen from the figure, after the handover is finished there is a

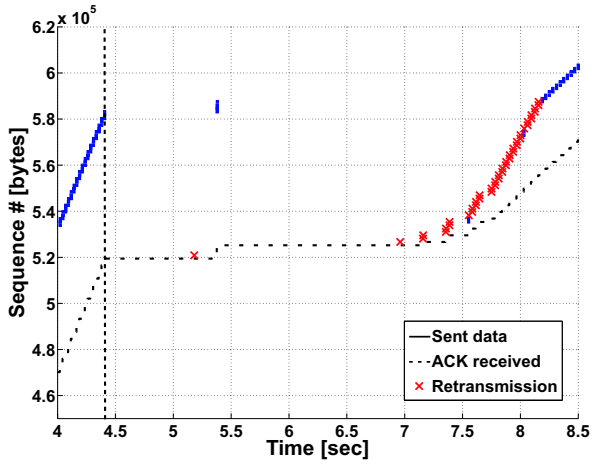


**Figure E.1:** *Time sequence graph: no handover delay, Eifel enabled, No buffering.*

**Table E.1:** *TCP traces illustrating timestamps spurious retransmission detection that leads to timeout.*

| | | | | | |
|---|---|---|---|---|---|
| | | | *Sender Side* | | |
| 1 | 4.453809 | <– | | ack 519425 | <timestamp 586403007/535598120> |
| 2 | 4.456609 | –> | 580241:581689(1448) | ack 1 | <timestamp 535598173/586403007> |
| 3 | 4.457840 | –> | 581689:583137(1448) | ack 1 | <timestamp 535598173/586403007> |
| 4 | 5.232270 | –> | 519425:520873(1448) | ack 1 | <timestamp 535598251/586403007> |
| 5 | 5.426615 | <– | | ack 525217 | <timestamp 586403098/535598125> |
| 6 | 5.429447 | –> | 583137:584585(1448) | ack 1 | <timestamp 535598270/586403098> |
| 7 | 5.430678 | –> | 584585:586033(1448) | ack 1 | <timestamp 535598270/586403098> |
| 8 | 5.431980 | –> | 586033:587481(1448) | ack 1 | <timestamp 535598270/586403098> |
| 9 | 5.623774 | <– | | ack 525217 | <timestamp 586403118/535598125> |
| 10 | 5.655029 | <– | | ack 525217 | <timestamp 586403121/535598125> |
| 11 | 5.686278 | <– | | ack 525217 | <timestamp 586403124/535598125> |
| 12 | 7.011958 | –> | 525217:526665(1448) | ack 1 | <timestamp 535598429/586403124> |
| | | | *Receiver Side* | | |
| 1 | 4.427181 | <– | 517977:519425(1448) | ack 1 | <timestamp 535598120/586402954> |
| 2 | 4.427265 | –> | | ack 519425 | <timestamp 586403007/535598120> |
| 3 | 4.439181 | <– | 519425:520873(1448) | ack 1 | <timestamp 535598123/586402957> |
| 4 | 4.451181 | <– | 520873:522321(1448) | ack 1 | <timestamp 535598123/586402957> |
| 5 | 4.451265 | –> | | ack 522321 | <timestamp 586403009/535598123> |
| 6 | 4.463181 | <– | 522321:523769(1448) | ack 1 | <timestamp 535598125/586402959> |
| 7 | 4.475180 | <– | 523769:525217(1448) | ack 1 | <timestamp 535598125/586402959> |
| 8 | 4.475274 | –> | | ack 525217 | <timestamp 586403012/535598125> |
| 9 | 5.338559 | <– | 519425:520873(1448) | ack 1 | <timestamp 535598251/586403007> |
| 10 | 5.338571 | –> | | ack 525217 | <timestamp 586403098/535598125> |
| 11 | 5.535728 | <– | 583137:584585(1448) | ack 1 | <timestamp 535598270/586403098> |
| 12 | 5.535740 | –> | | ack 525217 | <timestamp 586403118/535598125> |
| 13 | 5.566978 | <– | 584585:586033(1448) | ack 1 | <timestamp 535598270/586403098> |

timeout and a retransmission occurs at around 5.2 seconds. An ACK is received a while after and instead of continuing with the retransmission of the other lost packets, a couple of new packets are transmitted at around 5.4 seconds. The reason for this behaviour is illustrated further in Table E.1 which shows the traces of the TCP packets captured at the sender and receiver just before and after the handover. Arrows pointing to the left and right indicate incoming and outgoing packets, respectively, at the machine where the packets are captured.

Just before the handover the receiver gets the packet with sequence number 523769, at time 4.475180 (receiver trace line #7), and it sends an ACK requesting the next packet at time instant 4.475274 (receiver trace line #8). However,

this ACK (and also the previous ACK) are dropped as they were in flight at the time of the handover. Thus, the last ACK received by the sender before the handover is at 4.453809 (sender trace line #1) and this ACK was requesting packet with sequence number of 519425. When the RTO expires at 5.232270 (sender trace line #4), the packet with sequence number 519425 is retransmitted, even though it was correctly received, as the ACK signifying that was lost. This retransmitted packet has a timestamp value of <535598251/586403007>, where the first part is the current time at the sender, and the second one is the timestamp of the last ACK received signifying the receipt of the packet just before the retransmitted packet.

The receiver gets this retransmission at time instant 5.338559 seconds (receiver trace line #9) and it sends an ACK requesting for packet number 525217 (receiver trace line #10), as that is the next packet expected at the receiver. Note that this ACK has the timestamp value <586403098/ 535598125>, where the first part is the current time at the receiver and the second one is the timestamp of the packet that generated this ACK. When the sender gets this ACK (sender trace line #5), it compares the replied timestamp (535598125) with the timestamp sent with the retransmission (535598251) and finding out that it is smaller, concludes that the retransmission of the concerned packet was spurious and leaves the recovery stage, restores the congestion window and starts sending new data.

Although the decision that the retransmission of that particular packet was spurious is right, it creates a problem because we have to wait for another timeout (at time instant 7.011958), and hence an extra idle time of 1.3 seconds, before we start retransmitting the packets that are actually lost.